
Meridian 1

X11 Release 25.40

Succession Communication Server for Enterprise 1000

X21 Release 2.0

IP Line

Description, Installation, and Operation

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Standard 1.00

Contents

About this Document	13
Subject	13
Applicable systems	13
Structure and conventions	13
Description	15
Contents	15
Reference list	16
Overview	17
Applicable systems	20
System requirements	20
System configurations	25
Loadware delivery	27
Required packages	27
IP Line package components lists	28
Voice Gateway Media Card description	35
Virtual superloops, virtual TNs, and physical TNs	46
Internet Telephone registration	48
Zones	52
Administration	53
IP Line Feature Enhancements	57
Contents	57

Reference list	58
Overview	58
Feature Enhancements	60
Capacity Engineering Guidelines	119
Contents	119
Reference list	119
Overview	120
Capacity engineering	120
Internet Telephone Engineering	128
Product compatibility with other IP products	131
IP Network Engineering Guidelines	133
Contents	133
Reference list	134
Overview	134
IP address requirements	135
IP network assessment procedure	138
Codecs	140
Calculate IP Line traffic requirements	155
Assess WAN link resources	161
VoIP bandwidth management zones	164
Relationship between zones and subnets	167
Set service parameters	171
IP Line ELAN and TLAN configuration	194
LAN Device Interface Names	198
Installation and Configuration Summary	199
Contents	199
Reference list	199
Overview	200

Meridian 1 Rel 25.30 or Succession CSE 1000 Rel 1.1 systems	200
Succession CSE 1000 Rel 2.0 systems	204
Voice Gateway Media Card installation summary sheet	208
Internet Telephone configuration data summary sheet	210

Configuration of the DHCP Server 211

Contents	211
Reference list	211
Overview	212
i2002 and i2004 Internet Telephone, and i2050 Software Phone	212
Configuring the DHCP server to support Full DHCP mode	215

Installation and Initial Configuration of an IP Telephony Node 223

Contents	223
Reference list	224
Overview	224
Equipment considerations	225
Install the hardware components	226
Initial configuration of IP Line data on Meridian 1 and Succession Communication Server for Enterprise 1000	247

Configuration of IP Telephony Nodes using OTM 2.0 267

Contents	267
Reference list	268
Overview	268
Configure IP Line data using OTM	269
Transmit node configuration from OTM to the Voice Gateway Media Cards	300
Upgrade the Voice Gateway Media Card loadware and Internet Telephone firmware	307

Configuring OTM Alarm Management to receive IP Line	
SNMP traps	323
Assemble and install an Internet Telephone	328
Change the default IPL> CLI Shell password	328
Configure the Internet Telephone Installer Passwords	328

Configuration of IP Telephony Nodes using Element Management 329

Contents	329
Reference list	330
Overview	331
Configure IP Line data using Element Management	332
Transfer IP Telephony node configuration from Element Management to the Voice Gateway Media Cards	368
Upgrade the Voice Gateway Media Card loadware and Internet Telephone firmware	377
Configure OTM alarm management to receive IP Line	
SNMP traps	402
Assemble and install an Internet Telephone	402
Change the default IPL> CLI Shell password	402
Configure the Internet Telephone Installer Passwords	403
Importing node configuration from an existing node	403

IP Line Application Administration 407

Contents	407
Reference list	408
Overview	408
Password Security	410
Reset the Operational Measurements	434
Display the number of DSPs	434
Display IP Telephony node Properties	434
Display Voice Gateway Media Card properties	436

Transfer files using the Command Line Interface	437
IP configuration commands	438
Configure TLAN parameters	439
Packet loss monitor	440
Download the IP Line error log	441
Lamp audit and Keep Alive functions	441
Administration of IP Line Feature Enhancements	443
IP Line Administration using OTM 2.0	449
Contents	449
Reference List	450
Overview	450
OTM administration procedures	451
Back up and restore OTM data	461
Updating IP Telephony node properties using OTM	462
Update Voice Gateway Media Card card properties	483
Adding an IP Telephony node in OTM by retrieving an existing node	490
IP Line Command Line Interface access using Telnet or local RS-232 maintenance port	494
IP Line Administration using Element Management	497
Contents	497
Reference list	498
Overview	498
Element Management administration procedures	498
Backup and restore data	506
Update IP Telephony node properties	510
Update other node properties	538
Telnet to a Voice Gateway Media Card	538
Checking the Voice Gateway Channels	541

Setting the Internet Telephone Installer Password	545
Voice Gateway Media Card Maintenance	551
Contents	551
Reference list	552
Overview	552
Faceplate maintenance display codes	552
System error messages	557
IP Line and Internet Telephone maintenance and diagnostics - LD 32	564
IP Line CLI commands	567
Voice Gateway Media Card self-tests	583
Troubleshooting a loadware load failure	583
Troubleshooting Internet Telephone Installation	588
Maintenance Telephone	588
Upgrading Voice Gateway Media Card firmware	589
Replacing the Succession Media Card's CompactFlash	595
Voice Gateway Media Card Maintenance using OTM 2.0	597
Contents	597
Reference list	597
Overview	598
Replacing a Voice Gateway Media Card	598
Accessing the IPL> Command Line Interface from OTM	605
Adding a "dummy" node for retrieving and viewing IP Telephony node configuration	605
Voice Gateway Media Card Maintenance using Element Management	613
Contents	613
Reference list	613

Overview	613
Replacing a Voice Gateway Media Card	614
Accessing the CLI commands within Element Management	621
Access the IPL> Command Line Interface from Element Management	631

Appendix A: I/O, Maintenance, and Extender Cable Description 633

Contents	633
Overview	633
NTMF04EA I/O cable	634
Connector pin assignments	636
NTAG81CA maintenance cable description	639
NTAG81BA maintenance extender cable	641
Replacing the NT8DS1BA cable with the NT8D1AA cable and installing the NTCW84JW special IPE filter	642

Appendix B: RM356 Modem Router 647

Contents	647
Overview	647
RM356 modem router security features	648
Install the RM356 modem router	650
Configure the RM356 modem router by the manager menu	651
RM356 modem router manager menu description	658

Appendix C: Product Integrity 667

Contents	667
Overview	667
Reliability	668
Environmental specifications	669
Electrical regulatory standards	671

Appendix D: Subnet Mask Conversion from CIDR to Dotted Decimal Format 675

Overview	675
----------------	-----

Appendix E: DHCP Supplementary Information . 677

Contents	677
Introduction to DHCP	678
IP Acquisition Sequence	683
Internet Telephone support for DHCP	688

Appendix F: Setup and Configuration of DHCP Servers 697

Contents	697
Installing a Windows NT 4 server	698
Configuring a Windows NT 4 server with DHCP	698
Configuring a Windows 2000 server with DHCP	704
Installing ISC's DHCP Server	710
Configuring ISC's DHCP Server	710
Installing and configuring a Solaris 2 server	716

Appendix G: Downloading IP Line files from Nortel Networks Web Site 719

Contents	719
Overview	719

About this Document

Subject

This document describes the physical and functional characteristics of the Meridian 1 and Succession Communication Server for Enterprise 1000 IP Line 3.0 application and its use on the Voice Gateway Media Cards. This document also explains how to engineer, install, configure, administer, and maintain an IP Telephony node that contains Voice Gateway Media Cards.

Applicable systems

This document applies to a Meridian 1 large system (Option 51C, 61C, or 81C), a Meridian 1 small system (Option 11C or 11C Mini), and a Succession Communication Server for Enterprise (CSE) 1000 system. For the purposes of this document, all these systems are referred to generically as "system."

Structure and conventions

This document has separate chapters which are applicable only to either Optivity Telephony Manager (OTM) or Succession CSE 1000 Element Management.

The configuration, administration, and maintenance sections are divided into three chapters each. For example, there is a generic configuration chapter dealing with tasks related to installing and configuring IP Line. This chapter is followed by two other configuration chapters, one for OTM and another for Element Management. The administration and maintenance chapters have the same format.

The "Installation and Configuration Summary" chapter contains a summary of the steps for installing the Meridian 1 Rel 25.30 and Succession CSE 1000 Rel 1.1 systems (see page 200) and the Succession CSE 1000 Rel 2.0 systems (see page 204).

Description

Contents

This section contains information on the following topics:

Reference list	16
Overview	17
Applicable systems	20
Unsupported products	20
System requirements	20
OTM and Element Management	23
System configurations	25
Succession CSE 1000 Rel 2.0 Normal configuration	25
Succession CSE 1000 Rel 2.0 Branch Office configuration	26
Meridian 1 Rel 25-40 and Succession CSE 1000 Rel 1.1 configuration	27
Loadware delivery	27
Required packages	27
IP Line package components lists	28
Meridian 1 and Succession CSE 1000 Rel 1.1 package components	28
Succession CSE 1000 Rel 2.0 package components	30
Succession CSE 1000 Rel 2.0 Branch Office package components	32
Documentation	34
Voice Gateway Media Card description	35
ITG-P Line Card controls, indicators, and connectors	38
Succession Media Card controls, indicators, and connectors	42
Functional description of the Voice Gateway Media Cards	44
Virtual superloops, virtual TNs, and physical TNs	46

Virtual TNs	47
Internet Telephone registration	48
Succession CSE 1000 Rel 2.0 registration	48
Meridian 1 and Succession CSE 1000 Rel 1.1 registration	49
Virtual Terminal Manager description	49
Interactions with Internet Telephones	49
Signaling and messaging	50
Signaling protocols	50
ELAN TCP Transport	51
Zones	52
Administration	53
OTM 2.0's IP Line 3.0 application	53
Element Management	54
Command Line Interface	54
Overlays	55

Reference list

The following are the references for this section:

- *Internet Terminals Description* (553-3001-217)
- *Element Management* (553-3023-222)

Overview

IP Line 3.0 application provides an interface between an Internet Telephone and the Meridian 1 PBX / Succession Communication Server for Enterprise (CSE) 1000 Call Server. The IP Line 3.0 application extends the functionality of the ITG Pentium (ITG-P) Line card and introduces the Succession Media Card, a 32-port card.

The new functionality of the IP Line 3.0 application includes:

- New i2002 Internet Telephone support
- New platform support for the Succession Media Card and Succession Signaling Server
- New user features
(see Table 1 on page 21 for a list of supported features)
- Additional administration and supportability

A Dynamic Host Configuration Protocol (DHCP) server can be used to provide the required information to enable the Internet Telephone network connection and connect to the Voice Gateway Media Card.

Note: An ITG-P Line Card or a Succession Media Card with the IP Line 3.0 application installed is known as a Voice Gateway Media Card.

The Internet Telephone uses the IP network to communicate with the Voice Gateway Media Card and the optional DHCP server. Figure 1 on page 18 shows a system block diagram of the Meridian 1 and Succession CSE 1000 Rel 1.1 system. Figure 2 on page 19 shows a diagram of the Succession CSE 1000 Rel 2.0 system.

Refer to the *Internet Terminals Description* (553-3001-217) for more information on the following Internet Telephones:

- i2002 Internet Telephone
- i2004 Internet Telephone
- i2050 Software Phone

Figure 1
Meridian 1 and Succession CSE 1000 Rel 1.1 System Architecture

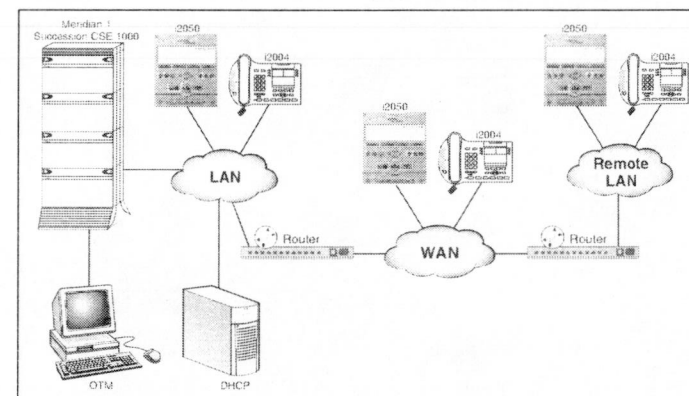
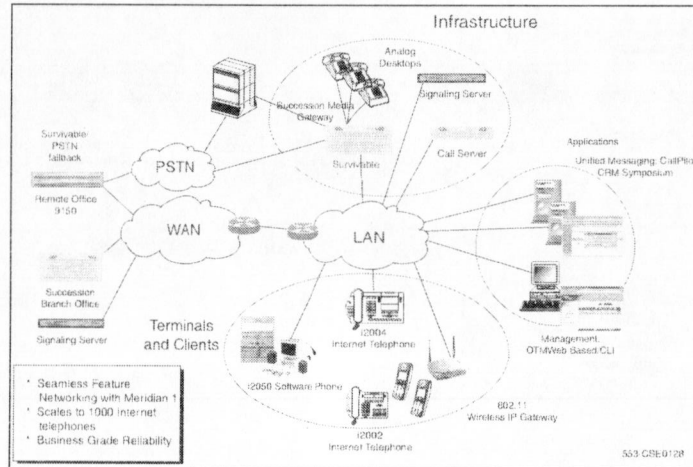


Figure 2
Succession CSE 1000 Rel 2.0 System Architecture



Applicable systems

The Meridian 1 and Succession CSE 1000 systems support the ITG-P Line Card and the Succession Media Card.

Unsupported products

The following remote service products do not support the ITG-P Line Card or Succession Media Card:

- Carrier Remote
- Mini-carrier Remote
- Fiber Remote
- Fiber Remote Multi-IPE

System requirements

Succession CSE 1000 Release 2.0 is the minimum system software that is required to have the complete functionality available in the IP Line 3.0 application loadware.

The IP Line 3.0 application is designed to run on Meridian 1 and Succession CSE 1000 systems. However, some of the new feature functionality requires changes to the PBX CPU software, as well as, to the IP Line application. As a result, the complete new feature functionality can be supported only on Succession CSE 1000 Release 2.0.

Some IP Line 3.0 application features are backward compatible with Meridian 1 Releases 25.15*, 25.30 and 25.40, and Succession CSE 1000 Release 1.1; however, not all the added functionality with Succession CSE 1000 Release 2.0 is available.

Note: * Meridian 1 Rel 25.15 is supported only in the following regions: Europe, Middle East, and Africa (EMEA).

Table 1 on page 21 outlines the new features available for Succession CSE 1000 Release 2 and also shows the features which are backward compatible with previous systems. Also refer to "IP Line Feature Enhancements" on page 57 for information about these new features.

Table 1
IP Line 3.0 new feature support (Part 1 of 2)

Feature	CSE 1000 Rel 2.0	CSE 1000 Rel 1.1	Meridian 1 Rel 25.40 Meridian 1 Rel 25.30 Meridian 1 Rel 25.15 ^a
Support of the i2002 Internet Telephone	Yes	No	No
Support for i2002/i2004 firmware version 1.3x	Yes	Yes (i2004 only)	Yes (i2004 only)
Succession Media Card Platform	Yes	Yes	Yes
Support for Signaling Server	Yes	No	No
NAT enhancement	Yes	Yes	Yes
Patching	Yes	Partial ^b	Partial ^b
802.1Q	Yes	Yes	Yes
Corporate Directory	Yes ^c	No	No
Data Path Capture tool	Yes	Yes	Yes
CSE Element Management support	Yes	No	No
Call statistics enhancements	Yes	No	No
User-defined Feature Key Labels	Yes	No	No
Private Zone	Yes	No	No
Graceful TPS Disable	Yes	Yes	Yes
Firmware download	Yes	Yes	Yes
Run-time download	Yes	Partial ^d	Partial ^d
Maintenance Audit Enhancement	Yes	Yes	Yes

Table 1
IP Line 3.0 new feature support (Part 2 of 2)

Feature	CSE 1000 Rel 2.0	CSE 1000 Rel 1.1	Meridian 1 Rel 25.40 Meridian 1 Rel 25.30 Meridian 1 Rel 25.15 ^a
Watchdog Timer	Yes	Yes	Yes
Improved Login Banner and Password Guessing Protection	Yes	Partial ^e	Partial ^e

- a. Meridian 1 Rel 25.15 is supported only in the following regions: Europe, Middle East, and Africa (EMEA).
b. Node level patching is not provided by OTM 2.0. The patching CLI command of the ITG-P Line Card and Succession Media Card can be used.
c. OTM 2.0 is required for the creation of the Corporate Directory database.
d. For more information, see "Run-time configuration changes" on page 82.
e. For more information, see "Login Banner Enhancement" on page 411 and "Password Guessing Protection" on page 412.

OTM and Element Management

Optivity Telephony Manager (OTM) 2.0 and CSE 1000 Element Management are used throughout this document as the primary interface for Voice Gateway Media Card and IP Line 3.0.

OTM 2.0 includes an application called "ITG Line 3.0" which is used to configure, administer, and maintain the IP Line on Meridian 1 and Succession CSE 1000 Release 1.1 systems. OTM 2.0 is the minimum required version.

CSE 1000 Element Management is the tool that is used as the configuration, administration, and maintenance interface for IP Line on the Succession CSE 1000 Release 2.0 system. CSE 1000 Element Management is referred to as Element Management for the remainder of this document.

Support of OTM's ITG Line 3.0 application

OTM's ITG Line 3.0 application fully supports IP Line 3.0 on the Meridian 1 Rels 25.15, 25.30, 25.40 and Succession CSE 1000 Rel 1.1 systems. OTM's ITG Line 3.0 application provides the feature set equivalent of the older ITG Line 2.2 application. IP Line 3.0 is not supported on systems prior to Release 25.15.

OTM's ITG Line 3.0 application provides limited functionality and support for Succession CSE 1000 Rel 2.0 systems. Instead, CSE 1000 Element Management is used for base line configuration of the Succession CSE 1000 Rel 2.0 systems. However, OTM 2.0 is required for Alarm Management and the Corporate Directory feature. OTM 2.0 also provides some Operational Measurement reporting.

Note: Element Management does have some Operational Measurement report capabilities.

OTM 2.0 is the only management tool for IP Line on Meridian 1 Rel 25.xx and Succession CSE Rel 1.1 systems. Element Management is the primary management tool for Succession CSE 1000 Rel 2.0; however, OTM 2.0 does provide partial support.

Table 2 outlines the systems that use OTM's ITG Line 3.0 application for the configuration, administration, and maintenance of IP Line.

Table 2
Systems using OTM's ITG Line 3.0 application for IP Line management

Supported Release		The following systems use OTM's ITG Line 3.0 application for IP Line 3.0
Meridian 1	19	Not supported
	20	Not supported
	21	Not supported
	22	Not supported
	23	Not supported
	24	Not supported
	25.15	Mandatory
	25.30	Mandatory
	25.40	Mandatory
CSE 1000	1.0	Not supported
	1.1	Mandatory
	2.0	Limited support and functionality* (*Uses Element Management for majority of IP Line configuration.)

System configurations

IP Line 3.0 can be used on different system configurations and its use varies on different system configurations. There are three different systems configurations:

- 1 Succession CSE 1000 Rel 2.0 Normal configuration
- 2 Succession CSE 1000 Rel 2.0 Branch Office configuration
- 3 Meridian 1 and Succession CSE 1000 Rel 1.1 configuration

Succession CSE 1000 Rel 2.0 Normal configuration

In the Normal configuration, there is a Succession Signaling Server in the system. The Signaling Server is a server that provides signaling interfaces to the IP network. The Signaling Server provides a central processor to drive the signaling for Internet Telephone and IP Peer Networking.

With IP Line 3.0, the Terminal Proxy Server (TPS) executes on the Signaling Server card and the media gateway executes on the ITG-P Line Card or the Succession Media Card (the Voice Gateway Media Cards). All Internet Telephones register with the Signaling Server card. The Voice Gateway Media Cards provide just the gateway media access. The Signaling Server also acts as the Master for the node.

A backup Signaling Server can exist in the Normal configuration. If the primary Signaling Server fails, the backup Signaling Server takes over and all Internet Telephones reregister with the backup.

If the primary Signaling Server has no backup and the primary Signaling Server fails, one of the Voice Gateway Media Cards is elected to be the node Master. The Internet Telephones then register to the Voice Gateway Media Cards. The same is true if the backup Signaling Server fails.

Multiple nodes can be configured in Succession CSE 1000 Release 2.0. If there are multiple nodes, the primary node is the Signaling Server (the Leader and Master).

If additional nodes are configured on the system, the Voice Gateway Media Cards in these nodes operate the same as in ITG Line 2.0-2.2. That is, the card provides both the TPS and the voice gateway functionality. In each node, one Voice Gateway Media Card is configured as a Leader.

In a Succession CSE 1000 Rel 2.0 system, the Signaling Server (if it exists) is the Leader. If additional Signaling Servers exist, they are known as followers. If the Leader Signaling Server fails and a follower Signaling Server exists, then the "follower" Signaling Server becomes the Leader. In the unlikely event that the new Leader Signaling Server also fails, the Leader role is transferred to one of the Voice Gateway Media Cards. An election is held and one of the Voice Gateway Media Cards becomes the Master. The display on that card's faceplate is Mxxx. When the Signaling Server comes back online, the mastership is transferred back to the Signaling Server. The Voice Gateway Media Card becomes a follower again and F000 is displayed on the card's faceplate.

Succession CSE 1000 Rel 2.0 Branch Office configuration

In the Branch Office configuration, the Internet Telephones register first with the Branch Office TPS, then are redirected to the main office's TPS.

If the connection to the main office is lost, the Internet Telephones register with the Branch Office TPS and continue to have service. The Internet Telephones have service because the telephones are configured with the node IP address of the Branch Office TPS.

For more information about this configuration, refer to the *Branch Office Guide* (553-3023-258).

Meridian 1 Rel 25.40 and Succession CSE 1000 Rel 1.1 configuration

In the Meridian 1 and Succession CSE 1000 Rel 1.1 configurations, there is no Signaling Server in the system. Each ITG-P Line Card and Succession Media Card functions as both a TPS and voice gateway. Since there is no Signaling Server in the system, the TPS functionality is on the card just as it is with ITG Line 2.0-2.2. In this configuration, one card is configured as the Leader and Internet Telephones register with individual Voice Gateway Media Cards.

Loadware delivery

IP Line 3.0 supports loadware delivery through the following formats:

- 1 CompactFlash
- 2 Signaling Server CD-ROM
- 3 Downloadable from the Nortel Networks Web site

Note: IP Line 3.0 loadware is no longer available through CD-ROM delivery.

The IP Line loadware and related documentation (such as *Readme First* documents) can be downloaded from the Nortel Networks Web site. See Appendix G on page 719 for details.

Required packages

The Internet Telephones require the software packages listed in Table 3.

Table 3
Required packages

Package	Package number
Digital Set Package (DSET)	88
Aries Terminal Package (ARIES)	170

Note: To configure the IP Line 3.0 in groups 5-7 on Option 81C, the Fiber Network (FIBN) software package #365 is required.

IP Line package components lists

Meridian 1 and Succession CSE 1000 Rel 1.1 package components

Table 4 lists the IP Line package components for the Meridian 1 and Succession CSE 1000 Rel 1.1 systems.

Table 4
Meridian 1 and Succession CSE 1000 Rel 1.1
IP Line 32-Port package components (Part 1 of 2)

Component	Code
Succession Media Card 32-Port - IP Line 3.0 Voice Gateway Systems Package for Meridian 1 and Succession CSE 1000 Rel 2.0, includes: • Succession Media Card 32-Port assembly (NTDU40BA) • Succession IP Line 3.0 Voice Gateway CompactFlash • ITG EMC Shielding Kit • Readme First Document • Shielded 50-pin to Serial/ELAN/TLAN adaptor • PC Maintenance cable • Succession IP Line 3.0 Voice Gateway NTP (CD-ROM) • ITG-specific Meridian 1 Backplane 50-pin I/O Panel Filter Connector (see Note)	NTDU41CA
Succession Media Card 32 Port assembly	NTDU40BA
ITG EMC Shielding Kit	NTVQ83AA
Shielded 50-pin to Serial/ELAN/TLAN Adaptor	A0852632
PC Maintenance cable	NTAG81CA

Table 4
Meridian 1 and Succession CSE 1000 Rel 1.1
IP Line 32-Port package components (Part 2 of 2)

Component	Code
ITG-specific Meridian 1 Backplane 50-pin I/O Panel Filter Connector (see Note)	NTCW84JA
Succession IP Line 3.0 Voice Gateway NTP (CD-ROM), includes: • <i>IP Line: Description, Installation and Operation</i> (553-3001-204) • <i>Internet Terminals Description</i> (553-3001-217) • <i>i2002 Internet Telephone User Guide</i> • i2002 Quick Reference Card • <i>i2004 Internet Telephone User Guide</i> • i2004 Quick Reference Card • <i>i2050 Software Phone User Guide</i>	NTDW81AD
Succession IP Line 3.0 Voice Gateway Internet Telephone User Guides & Quick Reference Guides (CD-ROM), includes • <i>i2002 Internet Telephone User Guide</i> • i2002 Quick Reference Card • <i>i2004 Internet Telephone User Guide</i> • i2004 Quick Reference Card • <i>i2050 Software Phone User Guide</i>	NTDW85AA
Note: The I/O panel filter connector is not required for Option 11C, Option 11C-Mini, or Succession CSE 1000.	

Succession CSE 1000 Rel 2.0 package components

Table 5 lists the IP Line package components for the Succession CSE 1000 Rel 2.0 system.

Table 5
Succession CSE 1000 Rel 2.0
IP Line 32-Port package components (Part 1 of 2)

Component	Code
Succession Media Card 32-Port - IP Line 3.0 Voice Gateway Systems Package for Succession CSE 1000 Rel 2.0, includes: • Succession Media Card 32-Port Assembly (NTDU40BA) • Succession IP Line 3.0 Voice Gateway CompactFlash • ITG EMC Shielding Kit • Readme First Document • Shielded 50-pin to Serial/ELAN/TLAN adaptor • Succession IP Line 3.0 Voice Gateway NTP (CD-ROM)	NTDU41BB
Succession Media Card 32 Port assembly	NTDU40BA
ITG EMC Shielding Kit	NTVQ83AA
Shielded 50-pin to Serial/ELAN/TLAN Adaptor	A0852632

Table 5
Succession CSE 1000 Rel 2.0
IP Line 32-Port package components (Part 2 of 2)

Component	Code
Succession IP Line 3.0 Voice Gateway NTP (CD-ROM), includes: • <i>IP Line: Description, Installation and Operation</i> (553-3001-204) • <i>Internet Terminals Description</i> (553-3001-217) • <i>i2002 Internet Telephone User Guide</i> • i2002 Quick Reference Card • <i>i2004 Internet Telephone User Guide</i> • i2004 Quick Reference Card • <i>i2050 Software Phone User Guide</i>	NTDW81AD
Succession IP Line 3.0 Voice Gateway Internet Telephone User Guides & Quick Reference Guides (CD-ROM), includes • <i>i2002 Internet Telephone User Guide</i> • i2002 Quick Reference Card • <i>i2004 Internet Telephone User Guide</i> • i2004 Quick Reference Card • <i>i2050 Software Phone User Guide</i>	NTDW85AA

Succession CSE 1000 Rel 2.0 Branch Office package components

Table 6 lists the IP Line package components for the Succession CSE 1000 Rel 2.0 Branch Office.

Table 6
Succession CSE 1000 Rel 2.0 Branch Office
IP Line 8-Port package components (Part 1 of 2)

Component	Code
Succession Media Card 8-Port - IP Line 3.0 Voice Gateway Systems Package for Succession CSE 1000 Rel 2.0 Branch Office, includes: • Succession Media Card 8-Port Assembly (NTDU40AA) • Succession IP Line 3.0 Voice Gateway CompactFlash • ITG EMC Shielding Kit • Readme First Document • Shielded 50-pin to Serial/ELAN/TLAN adaptor • Succession IP Line 3.0 Voice Gateway NTP (CD-ROM)	NTDU41AB
Succession Media Card 8 Port assembly	NTDU40AA
ITG EMC Shielding Kit	NTVQ83AA
Shielded 50-pin to Serial/ELAN/TLAN Adaptor	A0852632

Table 6
Succession CSE 1000 Rel 2.0 Branch Office
IP Line 8-Port package components (Part 2 of 2)

Component	Code
Succession IP Line 3.0 Voice Gateway NTP (CD-ROM), includes: • <i>IP Line: Description, Installation and Operation</i> (553-3001-204) • <i>Internet Terminals Description</i> (553-3001-217) • <i>i2002 Internet Telephone User Guide</i> • i2002 Quick Reference Card • <i>i2004 Internet Telephone User Guide</i> • i2004 Quick Reference Card • <i>i2050 Software Phone User Guide</i>	NTDW81AD
Succession IP Line 3.0 Voice Gateway Internet Telephone User Guides & Quick Reference Guides (CD-ROM), includes • <i>i2002 Internet Telephone User Guide</i> • i2002 Quick Reference Card • <i>i2004 Internet Telephone User Guide</i> • i2004 Quick Reference Card • <i>i2050 Software Phone User Guide</i>	NTDW85AA

Documentation

The following documents are available on the Succession IP Line 3.0 Voice Gateway NTP CD-ROM and on the Nortel Networks Web site:

- *IP Line: Description, Installation and Operation* (553-3001-204)
- *Internet Terminals Description* (553-3001-217)
- *i2002 Internet Telephone User Guide*
- i2002 Quick Reference Card
- *i2004 Internet Telephone User Guide*
- i2004 Quick Reference Card
- *i2050 Software Phone User Guide*

The documents are also available from SERL (Succession Electronic Reference Library).

Voice Gateway Media Card description

Voice Gateway Media Card is a term used to encompass both the ITG-P Line Card and the Succession Media Card. These cards plug into an Intelligent Peripheral Equipment (IPE) shelf in the Meridian system and into a Media Gateway and Media Gateway Expansion in the Succession CSE 1000 Rel 2.0 system.

The ITG-P Line card (NTVQ55AA) occupies two slots while the Succession Media Card (NTVQ01BA) occupies only one slot. The Succession Media Card comes in two versions: 8-port and 32-port.

The Succession Media Card introduces the following features:

- It increases the packet processing power compared to that of the ITG-P Line card.
- It increases the channel density from 24 to 32 ports (for 32-port version).
- It reduces the slot count from a dual IPE slot to a single IPE slot.
- It supports up to 128 Internet Telephones for 32-port version, while 32 Internet Telephones are supported on the 8-port version.

The 8-port version is typically intended for the Branch Office configuration. The 8-port Succession Media Card can be upgraded to the 32-port version using a DSP card installation.

Table 7 on page 36 provides a comparison of the ITG-P Line Card and Succession Media Cards. The tables also shows a comparison of the 8-port and 32-port Succession Media Card.

Table 7
Comparison of ITG-P Line Card and Succession Media Card

Item	ITG-P Line Card	Succession Media Card (32 port)	Succession Media Card (8 port)
Total DSP Channels	24	32	8
Number of slots the card occupies	2	1	1
Operating System	VxWorks 5.3	VxWorks 5.4	VxWorks 5.4
Processor	Pentium	IXP1200	IXP1200
DSP	8 x TI5409	4 x TI5421	1 x TI5421
Telogy version	7.01	8.1 High Density version (8 ports for each DSP)	8.1 High Density version (8 ports for each DSP)
Number of Internet Telephones on each Voice Gateway Media Card	96	128	32
Image file name prefixes shown by swVersionShow command	IPL P	IPL SA	IPL SA
/C: drive	On board Flash 2 x 4Mb	Plug-in CompactFlash 16Mb	Plug-in CompactFlash 16Mb
Upgrade	Two images files	One image file (no backup)	One image file (no backup)

Voice Gateway Media Cards have an ELAN management Ethernet port (10BaseT) and a TLAN VoIP Ethernet port (10/100BaseT) on the I/O panel.

Note: ELAN or Embedded LAN is for isolation of critical telephony signaling between the Call Server and the other components. ELAN is also known as the Management LAN. TLAN or Telephony LAN is for telephony / voice / signaling traffic. The TLAN connects to the customer network and the rest of the world. TLAN is also known as the Voice LAN.

There is an RS-232 Maintenance Port connection on the faceplates of both the ITG-P Line Card and the Succession Media Card. The ITG-P Line Card has an alternative connection to the same serial port on the I/O backplane.



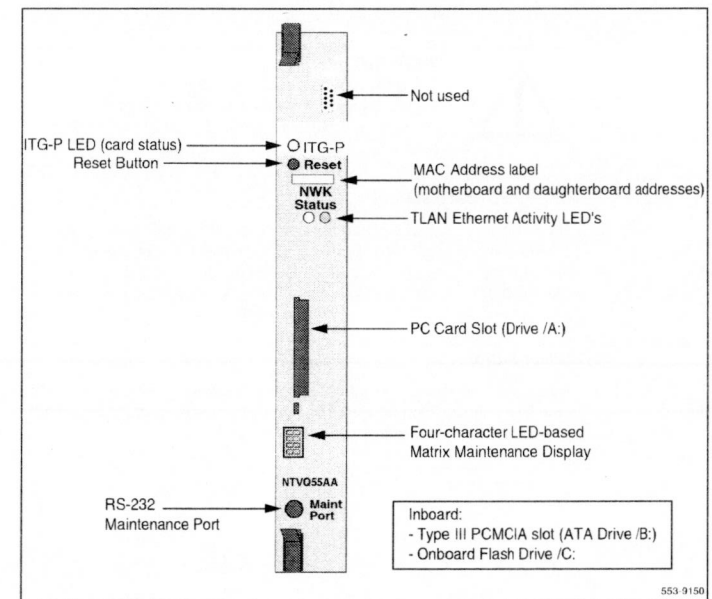
CAUTION

Do not connect maintenance terminals to both the faceplate and the I/O panel serial maintenance port connections at the same time.

ITG-P Line Card controls, indicators, and connectors

Figure 3 on page 38 shows the ITG-P Line Card faceplate components. The information in this section describes the components.

Figure 3
ITG-P Line Card (NTVQ55AA) assembly



553 9150

Faceplate components

The components on the faceplate of the ITG-P Line Card are described in the following sections.

NWK

The faceplate connector labeled NWK is a 9-pin, sub-miniature D-type connector. The connector is not used for the IP Line application.

**WARNING**

The NWK connector looks like a 9-pin serial connector. **DO NOT** connect a serial cable or any other cable to it. If you install a cable to the NWK connector, the TLAN is disabled.

ITG-P LED (card status)

The red status faceplate LED indicates the enabled/disabled status of the 24 card ports. The LED is on (red) during the power-up or reset sequence. The LED remains lit until the card is enabled by Meridian 1 or Succession CSE 1000. If the LED remains on, the self-test failed, the card is disabled, or the card rebooted.

Reset Button

Press the Reset switch to reset the card without having to cycle power to the card. This switch is normally used after a card loadware upgrade to the card or to clear a fault condition.

MAC Address label

The MAC Address label on the card's faceplate shows the motherboard and daughterboard addresses. The ELAN address corresponds to the Management MAC address. The Management MAC address for each card is assigned during manufacturing and is unchangeable. The ELAN/Management MAC address is the MOTHERBOARD Ethernet address found on the label. The MAC Address label on the ITG-P Line Card is similar to the following:

```
ETHERNET ADDRESS
MOTHERBOARD
00:60:38:8c:03:d5
DAUGHTERBOARD
00:60:38:01:b3:eb
```

TLAN Ethernet Activity LEDs (labeled NWK Status LEDs)

The NWK Status LEDs display the TLAN Ethernet activity.

- Green - The LED is on if the carrier (link pulse) is received from the TLAN Ethernet switch.
- Yellow - The LED flashes when there is TLAN data activity. During heavy traffic, the yellow LED can stay continuously lit.

Note: There are no Ethernet status LEDs for the ELAN management interface.

PC Card Slots

The ITG-P Line Card has one faceplate PC Card slot (designated Drive /A:). It is used for optional maintenance. The ITG-P Line Card also has one unused inboard slot (designated Drive /B:). The PC card slots support high-capacity PC flash memory cards.

Matrix Maintenance Display

A four-character, LED-based dot matrix display shows the maintenance status fault codes and other card state information.

RS-232 Maintenance Port (Maint Port)

The ITG-P Line Card faceplate provides a female 8-pin mini-DIN serial maintenance port connection (labeled Maint Port). An alternative connection to the faceplate serial maintenance port exists on the NTMF94EA I/O panel breakout cable.

**CAUTION**

Do not connect maintenance terminals or modems to the faceplate and I/O panel DB-9 male serial maintenance port at the same time.

Backplane interfaces

The backplane connector provides ELAN, TLAN, alternate connection to the serial maintenance port DS-30X and Card LAN interfaces.

DS-30X voice/signaling

DS30X carries Pulse Code Modulation (PCM) voice and proprietary signaling on the IPE backplane between the ITG-P Line Card and the Intelligent Peripheral Equipment Controller (XPEC).

Card LAN

Card LAN carries card polling and initialization messages on the IPE backplane between the ITG-P Line Card and the Intelligent Peripheral Equipment Controller (XPEC).

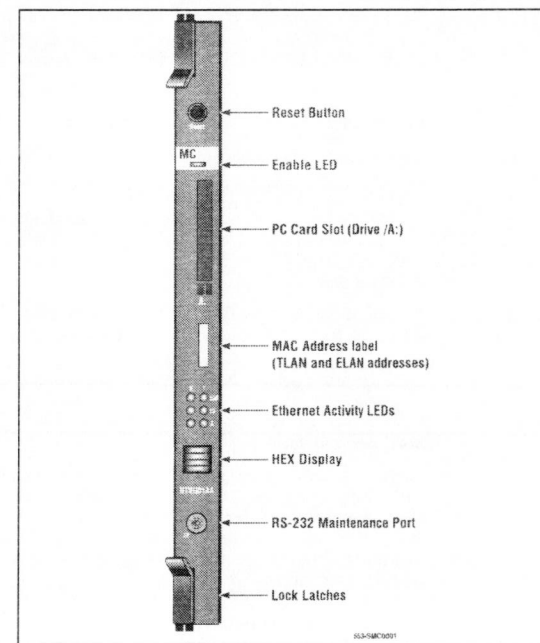
Assembly description

The ITG-P Line Card assembly is a two-slot motherboard and daughterboard combination. A PCI interconnect board connects the ITG motherboard and the DSP daughterboard.

Succession Media Card controls, indicators, and connectors

Figure 4 on page 42 shows the Succession Media Card faceplate.

Figure 4
Succession Media Card (NTVQ01BA or NTVQ01AA) assembly



Faceplate components

The components on the faceplate of the Succession Media Card are described in the following sections.

Reset Button

Use the Reset button on the faceplate to manually reset the Succession Media Card. This enables the card to be reset without cycling power to it. The Reset button is used to reboot the card after a loadware upgrade or to clear a fault condition.

Enable LED

The faceplate red LED indicates the following:

- the enabled/disabled status of the card
- the self-testing result during power up or card insertion into an operational system

PC Card Slot

This slot accepts the Type I or Type II standard PC Flash Cards, including ATA Flash cards (3 Mb to 170 Mb). The slot is labeled /A:.

Nortel Networks supplies PC Card adaptors that enable CompactFlash cards to be used in the slot.

MAC Address label

The MAC Address label on the card's faceplate is labeled ETHERNET ADDRESS. It shows the TLAN and ELAN addresses. The ELAN address corresponds to the card's Management MAC address. The Management MAC/ELAN address for each card is assigned during manufacturing and is unchangeable. The MAC Address label on the Succession Media Card is similar to the following:

ETHERNET ADDRESS
TLAN
00:60:38:BD:C9:9C
ELAN
00:60:38:BD:C9:9D

Ethernet Activity LEDs

The faceplate contains six Ethernet activity LEDs, three for the ELAN network and three for the TLAN network. The LEDs indicate the following links on the ELAN and TLAN (in order from the top):

- 1 100 (100BaseT)
- 2 10 (10BaseT)
- 3 A (Activity)

Maintenance Hex display

This is a four-digit LED-based hexadecimal display that provides the role of the card. It also provides an indication of fault conditions and the progress of PC card-based loadware upgrades or backups.

RS-232 Maintenance Port

The Succession Media Card faceplate provides a female 8-pin mini-DIN serial maintenance port connection. The faceplate on the card is labeled J2.

Functional description of the Voice Gateway Media Cards

The ITG-P Line Card and Succession Media Card can perform the following two separate functions depending on the system in which the card is located:

- It acts as a gateway between the circuit-switched voice network and the IP network.
- It acts as Terminal Proxy Server (TPS) or "virtual line card" for the i2002/i2004 Internet Telephones and i2050 Software Phone.

Gateway functional description

The Gateway:

- registers with the PBX using the TN Registration messages
- accepts commands from the PBX to connect/disconnect audio channel
- uses RTP/RTCP protocol to transport audio between the gateway and the Internet Telephone
- encodes/decodes audio from PCM to and from the Internet Telephone's format
- provides echo cancellation for the speaker on the i2002 and i2004 Internet Telephones (not applicable to the i2050 Software Phone)

Meridian 1 Rel 25.30 and Succession CSE 1000 Rel 1.1

Each ITG-P Line Card and Succession Media Card functions as both the LTPS and Voice Gateway, just as it did with ITG Line 2.0-2.2.

The Gateway portion of the card connects to the Meridian 1 or Succession CSE 1000 Rel 1.1 through the DS30X backplane. The Gateway portion also receives call speech path setup and codec selection commands through the ELAN port. The Internet Telephone connects to both the Gateway and the TPS functions through the TLAN port.

Succession CSE 1000 Rel 2.0

A Signaling Server is always present in the system. The LTPS executes on the Signaling Server card and the media gateway executes on the ITG-P Line Card or Succession Media Card. The Voice Gateway Media Cards provide only the gateway media access.

Virtual superloops, virtual TNs, and physical TNs

Virtual TNs (VTNs) enable configuration of service data for an Internet Telephone, such as key layout and class of service, without requiring the Internet Telephone to be dedicated (hard-wired) to a given TN on the Meridian 1 and Succession CSE 1000 Voice Gateway Media Card.

Calls are made between an Internet Telephone and circuit-switched telephone/trunks using the full Meridian 1 and Succession CSE 1000 feature set. Digital Signal Processor (DSP) channels are allocated dynamically for this type of call to perform the encoding/decoding required to connect the Internet Telephone to the circuit-switched network.

To create an Internet Telephone using VTNs, you must create a virtual superloop in LD 97.

- Up to 1024 VTNs can be configured on a single virtual superloop for Option 51C/61C/81/81C.
- Up to 128 VTNs can be configured on a single virtual superloop for Option 11C/11C-Mini, leading to a maximum number of 640 VTNs for each Option 11C/11C-Mini.
- Up to 128 VTNs can be configured on a single virtual superloop for Succession CSE 1000 Rel 1.0 and Rel 1.1
- Up to 1024 VTNs can be configured on a single virtual superloop for Succession CSE 1000 Rel 2.0. Table 8 on page 47 describes the virtual superloop and virtual card mapping on a Succession CSE 1000 Rel 2.0 system. Each superloop has two ranges of cards.

Table 8
Virtual superloop/virtual card mapping for Succession CSE 1000
Rel 2.0

SUPL	Card	
96	61-64	81-84
100	65-68	85-88
104	69-72	89-92
108	73-76	93-96
112	77-80	97-99

Each ITG-P Line card provides 24 physical TNs and the Succession Media Card provides 32 physical TNs. The physical TNs are the gateway channels (DSP ports). Configure the physical TNs (IPTN) in LD 14. They appear as tie trunks without a route data block.

Virtual TNs

Virtual TNs enable you to configure service data for a terminal, such as key layout and class of service, without requiring a physical terminal to be directly connected to the PBX/Call Server.

The concentration of Internet Telephones is made possible by dynamically allocating a port (also referred to as a physical TN) of the Voice Gateway Media Card for a circuit-switched to Internet Telephone call. All Meridian 1 and Succession CSE 1000 speech path management is done with physical TNs instead of the virtual TNs.

The channels (ports) on the Voice Gateway Media Cards are pooled resources.

The Internet Telephones (virtual TNs) are defined on virtual superloops.

A virtual superloop is a hybrid of real and phantom superloops. Like phantom superloops, no hardware (for example, XPEC or line card) is used to define and enable units on a virtual superloop. As with real superloops, virtual superloops use the time slot map to handle Internet Telephone (virtual TNs) to Internet Telephone calls.

Internet Telephone registration

The Terminal Proxy Server (TPS) maintains a count of the number of telephones registered to the card. Each IP Telephony node has one active Master. The active Master broadcasts to all Voice Gateway Media Cards and requests a response if it has room for another telephone.

The Election function uses a selection process to determine the node's Master. The Census function determines the Voice Gateway Media Cards within an IP Telephony node.

The maximum number of telephones for each ITG-P Line Card is 96 and the maximum number of telephones for each Succession Media Card is 128.

Note: The Succession Media Card is available in two versions: an 8-port card or a 32-port card. If the Succession Media Card has the 8-port configuration (that is, the DSP daughterboard is not installed), then the maximum number of telephones is 32.

Succession CSE 1000 Rel 2.0 registration

With Succession CSE 1000 Rel 2.0, the Internet Telephones register with the TPS on the Signaling Server. If a secondary Signaling Server exists, the Internet Telephone registrations are split between the primary and secondary Signaling Servers to aid in load balancing. In this case, the Internet Telephone registrations alternate between the primary and secondary Signaling Servers.

If the primary Signaling Server fails, the secondary Signaling Server takes over (if it exists) and the Internet Telephones register with the TPS on the secondary Signaling Server. If the secondary Signaling Server does not exist or in the case of failure of the secondary Signaling Server, the Internet Telephones register with the TPS functionality on the Voice Media Gateway Cards.

For more information on Signaling Server failure and redundancy, see *Planning and Engineering Guidelines* (553-3023-102).

Meridian 1 and Succession CSE 1000 Rel 1.1 registration

With Meridian 1 and Succession CSE 1000 Rel 1.1, the Internet Telephones register with the TPS on the individual Voice Gateway Media Cards (as there is no Signaling Server within these systems). The Voice Gateway Media Cards (ITG-P Line Cards and Succession Media Cards) include both the TPS and the voice gateway functionality.

If a Voice Gateway Media Card has Internet Telephones registered on it and the card fails, the Internet Telephones re-register with another Voice Gateway Media Card.

Virtual Terminal Manager description

The Virtual Terminal Manager (VTM):

- Arbitrates application access to the Internet Telephones.
- Manages all the telephones between the applications and the stimulus messaging to the telephone.
- Maintains context-sensitive states of the telephone (for example, display or lamp state).
- Isolates telephone-specific information from the applications (for example, the number of display lines, number of characters for each display line, tone frequency, and cadence parameters).

Interactions with Internet Telephones

The Internet Telephone receives the IP address of the Connect Server through either DHCP or manual configuration. The Internet Telephone contacts the Connect Server that instructs the Internet Telephone to display a message on its display screen requesting the customer's IP Telephony node number and TN.

After the node number and TN are entered, the Internet Telephone contacts the Node Master which selects a TPS with sufficient capacity to register the Internet Telephone. The chosen TPS contacts the Internet Telephone and, if the Internet Telephone is valid, registers it with the Meridian 1 or Succession CSE 1000. The registration information is then saved to the Internet Telephone.

Unregistration

If the Voice Gateway Media card detects a loss of connection with one of its registered Internet Telephones, it logs the event and sends an unregister message to the Meridian 1 or Succession CSE 1000 for that Internet Telephone.

Signaling and messaging

The IP Line application sends Scan and Signaling Distribution (SSD) messages through the Meridian 1 or Succession CSE 1000 ELAN. When tone service is provided, it is signaled to the TPS using new SSD messages sent through the ELAN.

Signaling protocols

The signaling protocol between the Internet Telephone and the IP Telephony node is the Unified Networks IP Stimulus Protocol (UNISim). The Reliable User Datagram Protocol (RUDP) is the transport protocol.

RUDP

RUDP is used for ELAN communications between the Meridian 1 or Succession CSE 1000 CPU and the Voice Gateway Media Cards, and for ELAN communications between the IP Telephony node and the Internet Telephones.

Signaling messages between the Voice Gateway Media Card and Internet Telephones use RUDP. Each RUDP connection is distinguished by its IP address and port number. RUDP is another layer on top of UDP. RUDP is proprietary to Nortel Networks.

The features of RUDP are:

- reliable communication system over a network
- packages are resent if an acknowledgement message (ACK) is not received following a time-out
- messages arrive in the correct sequence

- duplicate messages are ignored
- loss of contact detection

When a data sequence is packetized and sent from source **A** to receiver **B**, RUDP adds a number to each packet header to indicate its order in the sequence.

- If the packet is successfully transmitted to **B**, **B** sends back an ACK to **A**, acknowledging that the packet has been received.
- If **A** receives no message within a configured time, it retransmits the packet.
- If **B** receives a packet without having first received its predecessor, it discards the packet and all subsequent packets, and a NAK (no acknowledge) message which includes the number of the missed packet is sent to **A**. **A** retransmits the missed packet and continues.

UNISim

The Unified Network IP Stimulus protocol (UNISim) is the single point of contact between the various server components and the Internet Telephone.

UNISim is the stimulus-based protocol used for communication between an Internet Telephone and a Terminal Proxy Server on the Voice Gateway Media Card.

ELAN TCP Transport

Although TCP is used for the signaling protocol between the Call Server and the Voice Gateway Media Card, RUDP remains for the Keep Alive mechanism for the link. This means RUDP messages are exchanged to maintain the link status between the Call Server and the Voice Gateway Media Card.

There is no change on the TLAN signaling mechanism. Internet Telephones continue to use the RUDP transport protocol to communicate with the Voice Gateway Media Card.

The TCP protocol enables messages to be bundled. Unlike the RUDP transport that creates a separate message for every signaling message (such as display updates or key messages), the TCP transport bundles a number of messages and sends them as one packet.

Handshaking is added to the Call Server and IP Line loadware so that the TCP functionality is automatically enabled. A loadware version check is performed by the IP Line application each time before it attempts to establish a TCP link with the Meridian 1 and Succession CSE 1000 CPU. If the version does not satisfy the minimum supported version (Meridian 1 Rel 25.40 or Succession CSE 1000 Rel 1.1), a RUDP link is used instead.

For Meridian 1 Rel 25.40 and Succession CSE 1000 Rel 1.1 systems, TCP transports messages, while RUDP establishes and maintains the link. However, for Meridian 1 Rel 25.15 or Rel 25.30, RUDP is used to maintain the link and all signaling.

Zones

To optimize IP Line traffic bandwidth use between different locations, the IP Line network is divided into "zones" representing different topographical areas of the network. All Internet Telephones and IP Line ports are assigned a zone number indicating the zone to which they belong.

When a call is made, the codecs used vary depending on which zone(s) the caller and receiver are in.

By default, when a zone is created in LD 117:

- codecs are selected to optimize voice quality (BQ - Best Quality) for connections between units in the *same* zone.
- codecs are selected to optimize voice quality (BQ - Best Quality) for connections between units in *different* zones.

Each zone can be configured to:

- optimize either voice quality or bandwidth usage for calls between users in that zone.
- optimize either voice quality or bandwidth usage within a zone and all traffic going out of a zone.

See "VoIP bandwidth management zones" on page 164.

For more information about zones, refer to the following:

- Shared and Private zones (see "Private Zone Configuration" on page 73)
- Zones and Virtual Trunks (see *Meridian 1 Integrated Telephony Gateway Trunk 2.0/ISDN Signaling Link: Description, Installation, and Operation* (553-3001-202))
- Zones and Branch Office (see *Branch Office* (553-3023-221))

Administration

The Voice Gateway Media Card is administered using multiple management interfaces including:

- a Graphical User Interface (GUI) provided by OTM 2.0 called ITG Line 3.0.
- a Command Line Interface (CLI).
- administration and maintenance overlays of Meridian 1 and Succession CSE 1000 Call Servers.
- a Web browser interface provided by Element Management. Element Management is used for administering Voice Gateway Media Cards in the Succession CSE 1000 Rel 2.0 systems.

Note: Element Management is not available for earlier systems (Meridian 1 and Succession CSE 1000 Rel 1.1).

OTM 2.0's IP Line 3.0 application

For Meridian 1 systems, OTM 2.0 is required for IP Line 3.0. OTM is used for tasks such as the following:

- creating a node
- adding Voice Gateway Media Cards to the node
- transmitting loadware to the Voice Gateway Media Cards
- upgrading loadware

- defining SNMP alarms
- selecting codecs

Element Management

The Element Management Web server is required for Succession CSE 1000 Rel 2.0. Element Management enables the IP Line 3.0 application to be configured from a Web browser interface.

The "CSE 1000 Element Management" Web interface is used to manage the IP Line 3.0 application.

The Element Management Web interface is divided into two categories:

- 1 CSE 1000 Element Management - used to manage the Call Server and IP Telephony nodes (IP Line).
- 2 Gatekeeper Element Management - used to administer network numbering plan for Network Connect Server that is used by network-wide Virtual Office and Branch Office.

Command Line Interface

The Command Line Interface (CLI) provides a text-based interface to perform some specific Signaling Server and Voice Gateway Media Card installation, configuration, administration, and maintenance functions. You can establish a CLI session by connecting a TTY or PC to the card serial port or Telnet through the ELAN or TLAN IP address.

In the case of an IP Telephony node with no Signaling Server, the CLI must be used to configure the Leader card of the IP Telephony node so that OTM and Element Management can communicate with the Leader card and the node.

For more information about the CLI commands, see page 567.

Overlays

The following sections describe the changes that have been made to the Overlays with the introduction of IP Line 3.0.

LD 11

A new subtype for the i2002 Internet Telephone is introduced in LD 11. Also, the new CLS for the new Virtual Office feature has been added. The CLS prompt includes the VOLA/VOLD (Virtual Office Login Allowed/Denied) and VOUA/VOUD (Virtual Office User Allowed/Denied) for Virtual Office. LD 11 needs to accept CRPA/CRPD class of service input for the Corporate Directory feature on the Internet Telephones.

There are new prompts and responses available with LD 11. See Table 61: "LD 11 Configure an Internet Telephone" on page 263.

LD 14

There are two minor changes to the administration of the Voice Gateway Media Cards in LD 14. See "Configure physical TNs (LD 14)" on page 253 for the new responses (XTRK and MAXU). See Table 55: "Configure physical TNs in LD 14" on page 253 for prompts and responses.

LD 20

No new prompts have been added to LD 20. For the TYPE prompt, the i2002 response is introduced as a new customer response.

LD 81

Two changes have been made to LD 81. The FEAT prompt prints for the i2002 Internet Telephone and the FEAT prompt accepts VOLA, VOLD, VOUA, and VOUD for the new Virtual Office feature.

LD 82

No new prompts have been added to LD 83 however the i2002 Internet Telephone is printed.

Note: For a complete listing of the changes to the software input/output prompts, see *What's New for Succession Communication Server for Enterprise 1000* (553-3023-015).

IP Line Feature Enhancements

Contents

This section contains information on the following topics:

Reference list	58
Overview	58
Feature Enhancements	60
Support of the i2002 Internet Telephone	60
Corporate Directory	62
CSE 1000 Element Management support	63
Enhanced Call Statistics	64
Programmable line (DN) / Feature Key Labels	71
Private Zone Configuration	73
Interworking with NAT	77
Run-time configuration changes	82
Patching tool	85
Virtual Office	90
Branch Office	92
802.1Q Support	93
Data Path Capture tool	100
i2002 and i2004 Internet Telephone Firmware	101
Graceful Disable	107
Maintenance Audit enhancement	111
Hardware Watchdog Timer	116
Codecs	118

Reference list

The following are the references for this section:

- *i2002 Internet Telephone User Guide*
- *Internet Terminals Description* (553-3001-217)
- *Installing and Configuring Optivity Telephony Manager* (553-3001-230)
- *Features and Services* (553-3001-306)
- *Branch Office* (553-3023-221)
- *Element Management* (553-3023-222)
- *Maintenance* (553-3023-510)

Overview

The IP Line 3.0 application introduces a number of new feature enhancements. Table 9 lists these new features and the systems that support the new feature.

Table 9
IP Line 3.0 new feature support (Part 1 of 2)

Feature	CSE 1000 Rel 2.0	CSE 1000 Rel 1.1	Meridian 1 Rel 25.40 Meridian 1 Rel 25.30 Meridian 1 Rel 25.15 ^a
Support of the i2002 Internet Telephone	Yes	No	No
Support for i2002/i2004 firmware version 1.3x	Yes	Yes (i2004 only)	Yes (i2004 only)
Succession Media Card Platform	Yes	Yes	Yes
Support for Signaling Server	Yes	No	No
NAT enhancement	Yes	Yes	Yes
Patching	Yes	Partial ^b	Partial ^b

Table 9
IP Line 3.0 new feature support (Part 2 of 2)

Feature	CSE 1000 Rel 2.0	CSE 1000 Rel 1.1	Meridian 1 Rel 25.40 Meridian 1 Rel 25.30 Meridian 1 Rel 25.15 ^a
802.1Q	Yes	Yes	Yes
Corporate Directory	Yes ^c	No	No
Data Path Capture tool	Yes	Yes	Yes
CSE Element Management support	Yes	No	No
Call statistics enhancements	Yes	No	No
User-defined Feature Key Labels	Yes	No	No
Private Zone	Yes	No	No
Graceful TPS Disable	Yes	Yes	Yes
Firmware download	Yes	Yes	Yes
Run-time download	Yes	Partial ^d	Partial ^d
Maintenance Audit Enhancement	Yes	Yes	Yes
Watchdog Timer	Yes	Yes	Yes
Improved Login Banner and Password Guessing Protection	Yes	Partial ^e	Partial ^e

- a. Meridian 1 Rel 25.15 is supported only in the following regions: Europe, Middle East, and Africa (EMEA).
b. Node level patching is not provided by OTM 2.0. The patching CLI command of the ITG-P Line Card and Succession Media Card can be used.
c. OTM 2.0 is required for the creation of the Corporate Directory database.
d. For more information, see "Run-time configuration changes" on page 82.
e. For more information, see "Login Banner Enhancement" on page 411 and "Password Guessing Protection" on page 412.

Feature Enhancements

Support of the i2002 Internet Telephone

The IP Line 3.0 application supports three telephones:

- the new i2002 Internet Telephone
- the i2004 Internet Telephone
- the i2050 Software Phone

The i2002 Internet Telephone is similar in appearance and functionality to the i2004 Internet Telephone; however, the i2002 has a smaller display and less feature keys.

Table 10 on page 61 shows a comparison of the i2002 and i2004 Internet Telephones.

For detailed information about the i2002 Internet Telephone, see the following Guides:

- *i2002 Internet Telephone User Guide*
- *Internet Terminals Description (553-3001-217)*

Table 10
Comparison of the i2002 and i2004 Internet Telephones

Feature	i2004 Internet Telephone	i2002 Internet Telephone
Display		
Display size and format	3 line display 24 characters on each line	1 line display of 24 characters
Information Line	3 * 24 characters	1 * 24 characters
Dedicated Data/Time field	Yes	No
Context Label field	Yes	No
Keys		
Soft Keys	4 soft keys, soft-labeling 7 characters long	4 soft keys, soft-labeling 6 characters long
Feature Keys	6 soft keys, soft-labeling 10 characters long	4 soft keys, soft-labeling 10 characters long
Other features		
Feature set	Based on the M2616 with select M3900 features added	Based on the M2616 with select M3900 features added
DHCP support	Yes	Yes
Transducers	Headset (HS) / Handset (HD) / Handsfree (HF)	Headset (HS) / Handset (HD) / Handsfree (HF)
Voice Codec support	G.711, G729A, G729AB, G.723.1 ^a	G.711, G729A, G729AB, G.723.1 ^a
Firmware download	Automatic firmware version checking and download	Automatic firmware version checking and download
3-port unmanaged Layer 2 switch for data and voice	Depending on the model, the switch can be added on externally or built-in	Built-in

a. The G.723.1 codec is supported only on Succession Communication Server for Enterprise (CSE) 1000 Rel. 2.0.

Corporate Directory

The new Corporate Directory feature is based on the M3900 telephone Corporate Directory feature and is available on the i2002 Internet Telephones, i2004 Internet Telephones, and the i2050 Software Phone.

The Corporate Directory database is created using OTM 2.0 and is generated from one of the following:

- the configured DN information from the Call Server
- the data from a corporate LDAP server

The database is downloaded and stored on the Call Server. It is then accessible to the Internet Telephones.

The Directory key on the telephone is used to access the directory, select a listing, and then dial a number from the Corporate Directory. The Navigation keys are used to refine the search within the Corporate Directory.

Corporate Directory is configured in LD 11. LD 11 needs to accept CRPA/CRPD class of service for the Internet Telephones (see "Corporate Directory: LD 11 configuration" on page 443).

In the Meridian 1 or Succession CSE 1000 Rel 1.1 configuration, the TPS is on the Voice Gateway Media Card. The ITG-P Line Card supports the registration of 96 Internet Telephones and the Succession Media Card supports the registration of 128 Internet Telephones. Both cards provide the Corporate Directory feature to all registered Internet Telephones.

In the Succession CSE 1000 Rel 2.0 Normal configuration, the TPS is on the Signaling Server. Each Corporate Directory entry requires 500 bytes of memory, therefore, with sufficient memory the Signaling Server can support Corporate Directory access from the same number of telephones that are registered.

For more information about the operation of the Corporate Directory feature, refer to:

- *Installing and Configuring Optivity Telephony Manager* (553-3001-230)
- *Internet Terminals Description* (553-3001-217).

CSE 1000 Element Management support

CSE 1000 Element Management is a new feature of the Succession CSE 1000 Rel 2.0 product that enables configuration of IP Line 3.0 using a Web browser.

With the introduction of Succession CSE 1000 Rel. 2.0, each Signaling Server is the host of a new Web server, CSE 1000 Element Management, that enables users to perform configuration, administration, and maintenance of the system components. CSE 1000 Element Management is a graphical Web interface and it provides a graphical alternative to the traditional Overlays and Command Line Interface. The interface is available to users running a Web browser on their computer. No special client software is required.

The Element Management Web server runs on each Signaling Server and the Signaling Server acts as a file server.

When a Web browser is opened and the IP address of the Signaling Server is entered, the Element Management interface is displayed. Element Management is then used to perform tasks such as configuring an IP Telephony node; checking and uploading loadware and firmware files; and retrieving the CONFIG.INI and BOOTP.TAB configuration files from the Call Server. The Voice Gateway Media Cards are notified to FTP the files from the Call Server.

OTM 2.0's Navigators incorporate links to each CSE 1000 Element Management Web server in a network.

Note: For the remainder of this document, CSE 1000 Element Management is referred to as Element Management.

For more information, refer to *Element Management* (553-3023-222).

Enhanced Call Statistics

IP Line 3.0 introduces an additional level of statistics information that can be collected.

LD 32

New LD 32 commands

Four new commands have been added to LD 32. The new commands are:

- ENCT CARDS L S C <customer>
- ECNT ZONE zoneNum <customer>
- ECNT NODE nodeNum
- ECNT SS hostName

Table 11 on page 65 describes these new commands.

Table 11
Additional LD 32 commands (Part 1 of 2)

Command	Description
ECNT CARD L S C <customer>	This command prints the number of Internet Telephones registered for the specified card. - If the <customer> parameter is specified, the count is specific to that customer. A card must be specified to enter a customer. Otherwise, the count is across all customers. - If no parameters are entered, the count is printed for all zones. A partial TN can be entered for the card (L or L S) which then prints the count per that parameter. A customer cannot be specified in this case. Example: <pre>ecnt card 81 << Card 81 >> Number of Register Ethersets: 5 Number of Unregistered Ethersets: 27</pre>
ECNT ZONE zoneNum <customer>	This command prints the number of Internet Telephones registered for the specified zone. - If <customer> parameter is specified, the count is specific to that customer. A zone must be specified to enter a customer. Otherwise, the count is across all customers. - If no parameters are entered, the count is printed for all zones. Example: <pre>ecnt zone 0 0 << Zone 0 Customer 0 >> Number of Register Ethersets: 4 Number of Unregistered Ethersets: 17</pre>

Table 11
Additional LD 32 commands (Part 2 of 2)

Command	Description
ECNT NODE nodeNum	This command prints the number of Internet Telephones registered for the specified node. If the nodeNum parameter is not entered, the count is printed for all nodes. Example: <pre>ecnt node 8765 << Zone 8765 >> Number of Register Ethersets: 3</pre>
ECNT SS hostName	This command prints the number of Internet Telephones registered for the specified Signaling Server. If hostName parameter is not entered, the count is printed for all signaling servers. Example: <pre>ecnt ss << Signaling Server: BVWAlphaFox IP 10.10.10.242>> Number of Register Ethersets: 1000</pre>

Error messages for the new LD 32 commands

Error messages are printed when invalid data is entered for these new commands. The messages include valuable information such as the correct ranges for the command parameters. See the following tables for the error messages:

- Table 12: "ECNT Card command error messages" on page 67.
- Table 13: "ECNT Zone command error messages" on page 67.
- Table 14: "ECNT Node command error messages" on page 68.
- Table 15: "ECNT SS command error message" on page 68.

Table 12
ECNT Card command error messages

Error	Error Message
Slot out of range error	Slot out of range. Range: [61-99]
Slot non-virtual loop error	Slot does not correspond to a virtual loop.
Slot not configured loop error	Slot corresponds to a virtual loop but it is not configured.
Customer out of range error	Customer out of range. Range: [0-31]
Customer not configured error	Customer does not exist.
Combination of invalid slot and invalid customer	Slot does not correspond to a virtual loop. Customer out of range. Range: [0-31]

Table 13
ECNT Zone command error messages

Error	Error Message
Zone out of range error	Zone out of range. Range: [0-255]
Zone not configured error	Zone not configured.
Customer out of range error	Customer out of range. Range: [0-31]
Customer not configured error	Customer does not exist.
Combination of invalid zone and invalid customer error	Zone not configured. Customer out of range. Range: [0-31]

Table 14
ECNT Node command error messages

Error	Error Message
Node out of range error	Node out of range. Range: [0-9999]
Node not configured error	Node not registered.

Table 15
ECNT SS command error message

Error	Error Message
SS not found in system error	Signaling Server <name> does not exist.

LD 2 system traffic report

A new system traffic report 16 in LD 2 is created on the Succession CSE 1000 Rel 2.0 system to add the printing of Internet Telephone data at the zone level. The data is printed for the following categories at the end of each collection period on a per zone basis (the counts are reset after the data is printed):

- Total inter/intra calls made
- Total inter/intra calls blocked
- Percent average inter/intra zone bandwidth used
- Percent maximum inter/intra zone bandwidth used
- Total inter/intra zone bandwidth threshold exceeded count

The "Total inter/intra zone bandwidth threshold exceeded count" prints the number of times a user configured bandwidth threshold was exceeded for the zone during the collection period. The existing LD 2 commands that are related to setting the system threshold are used with a new value defined for the bandwidth threshold.

Table 16
System threshold commands

Command	Description
TTHS TH tv	Prints the current system thresholds.
STHS TH tv -- TV	Sets the system thresholds.
Note 1: The system thresholds TH values 1-4 already exist. A new TH value of 5 is used for the zone bandwidth threshold. Note 2: The system thresholds TV value is the percentage of the zone's maximum bandwidth. The range values are 000-999, where 000 corresponds to 00.0% and 999 corresponds to 99.9%. The default is 90.0%.	

The following examples first set the system bandwidth to 75% and then prints the actual value.

```
.STHS 5 750
.TTHS 5
```

The system traffic report 16 outputs data in the following format:

```
zone cmi cmo cbi cbo pi po ai ao vi vo
```

Table 17 describes the output data.

Table 17
System traffic report 16 data output

Data	Description
zone	Number of the zone
cmi	Intrazone calls made
cmo	Interzone calls made
cbi	Intrazone calls blocked
cbo	Interzone calls blocked
pi	Intrazone peak bandwidth (%)
po	Interzone peak bandwidth (%)
ai	Intrazone average bandwidth usage (%)
ao	Interzone average bandwidth usage (%)
vi	Intrazone threshold violations
vo	Interzone threshold violations

An example of the printout is:

```
.invs 16
0000 TFS016

000 00005 00003 00000 00000 07 03 02 01 000 000
001 00003 00003 00000 00000 03 2 01 01 000 000
```


All other commands (SOPS, COPS, TOPS) function as normal. Table 18 shows the SOPS, COPS, and TOPS commands:

Table 18
SOPS, COPS, TOPS commands

.tops 1 2 3 4 5 14	Displays the current system report list
.sops 1 2 3 4 5 14 -- 16	Add report 16 to be printed
.tops 1 2 3 4 5 14 16	Display system report list with report 16 added
.cops 1 2 3 4 5 14 16 -- 16	Delete report 16
.tops 1 2 3 4 5 14	Display system report list with report 16 deleted

Programmable line (DN) / Feature Key Labels

IP Line 3.0 gives the Internet Telephone user the ability to program the label on the feature key. This label change is saved and then displayed on the feature key. Feature keys are available on the i2002 (four feature keys) and i2004 Internet Telephones (six feature keys) and the i2050 Software Phone.

Prior to IP Line 3.0, a user could program a feature key for calling a certain phone number, for example, their home phone number. The label on the feature would be displayed as the phone number, for instance 555-1234. With the new IP Line 3.0 programmable line (DN) / Feature Key Labels, the user can change the label to say "Home" instead of their home phone number. The label can be up to a maximum of 10 characters.

The Feature Key labels for each Internet Telephone are stored in a text file in the `c:/a/db/database.rec` directory on the Call Server. The label information is retrieved from the file during the sysload of the Call Server into memory. When the Call Server performs EDD, the information is dumped to the file.

When the telephone registers, the Call Server looks up the Feature Key label in the memory based on the TN of the Internet Telephone. If the labels are found, they are sent to the telephone through SSD messages when the key map download occurs. If the labels are not found, the Call Server sends out the key number strings or key functions.

System impact

The Feature Keys (self-labeled) use space on the Call Server's /C: drive (flash or hard drive) to store its database and needs a structure to hold the information in memory. A system must have enough space in the memory and flash drive to store the database.

This feature introduces new messages into the network. A single SSD message can transfer only two characters. At the key map download time, it needs a maximum 60000 SSD messages from the Call Server to IP Line.

$60000 = (5 \times 12 \times 1000)$ where:

- 5 SSD messages to send one label
- 12 feature keys (6 keys on 2 feature key pages)
- 1000 Internet Telephones on the Signaling Server

When the user changes the feature key label, SSD messages are sent from IP Line to the Call Server. The Call Server has to save this information to memory and as a result, this information can impact the performance of the Call Server.

For more information about programmable line (DN)/feature keys (self-labeled), refer to *Internet Terminals Description* (553-3001-217).

Private Zone Configuration

DSP resources for each customer are placed in one common pool. A DSP channel is allocated to an IP to circuit-switched call based on a round-robin searching algorithm within the pool.

If an available resource cannot be found, the overflow tone is given. For most installations, this approach works because all Internet Telephone users share the IP Line DSP resources. The DSPs can be provisioned using a DSP-to-Internet Telephone ratio similar to trunk resources since the DSPs are used only for circuit-switched access or conference calls.

When IP to PSTN calls are used, such as with ACD agents or other users who consistently are using trunk resources when making calls, it becomes difficult to provision the system in a way that guarantees an available DSP channel when these users need it.

If the other users suddenly make a lot of conference calls or trunk calls, the DSP resources can deplete and as a result, calls cannot be made. This occurs because all DSP channels are in one pool. To address this situation, the IP Line 3.0 application adds the new Private Zone Configuration feature for DSP configuration and allocation. This new feature enables the configuration of one or more gateway channels as a private resource to guarantee DSP availability for critical or ACD agent Internet Telephones.

A new Private Zone classification has been added to the zone configuration. A zone can now be configured as shared or private.

Shared Zone

The current default zone type is a shared zone. The Internet Telephones configured in shared zones use DSP resources configured in shared zones. If all the shared zones' gateway channels are used, the caller receives an overflow tone and the call is blocked. Select gateway channels in the following order:

- Select a channel from the same zone as the Internet Telephone is configured.
- Select any available channel from the shared zones' channels.

Private Zone

The private zone is the new zone type introduced in IP Line 3.0. DSP channels configured in a private zone are used only by the Internet Telephones that have also been configured for that private zone. If more DSP resources are required by these Internet Telephones than what is available in the zone, DSPs from other shared zones are used.

Internet Telephones configured in shared zones cannot use the private zones' channels.

Select the gateway channels in the following order:

- Select a channel from the same private zone as the Internet Telephone is configured.
- Select any available channel from the pool of shared zones' channels.

LD 117

DSP channels and Internet Telephones are set as shared or private based on zone configuration. A new parameter was added to the zone configuration commands in LD 117. Zone configuration can be set to either shared or private using the parameter <zoneResourceType>.

A zone is configured in LD 117 as follows:

```
NEW ZONE <zoneNumber> [<intraZoneBandwidth>
<intraZoneStrategy> <interZoneBandwidth> <interZoneStrategy>
<newResourceType>]
```

```
CHG ZONE <zoneNumber> [<intraZoneBandwidth>
<intraZoneStrategy> <interZoneBandwidth> <interZoneStrategy>
<newResourceType>]
```

By default, a zone is configured as shared (newResourceType=shared).

Example

The command to add a new zone, zone 10, is:

```
new zone 10
```

```
Zone 10 added. Total number of Zone = n
(where n is the total number of zones)
```

The **prt zone** command is used to see details for all configured zones. See Table 19 for sample output of the **prt zone** command.

Table 19
Sample output from prt zone command

			Intrazone				Interzone				
Zone	State	Type	Bandwidth (Kbps)	Strategy	Usage (%)	Peak (Kbps)	Bandwidth (Kbps)	Strategy	Usage (%)	Peak (Kbps)	HO/BRCH
0	ENL	SHARED	100000	BQ	0	0	100000	BQ	0	0	HO
1	ENL	SHARED	10000	BQ	0	0	10000	BQ	0	0	HO
4	ENL	PRIVATE	10000	BQ	0	0	10000	BQ	0	0	HO
10	ENL	SHARED	10000	BQ	0	0	10000	BQ	0	0	HO

Function of the shared and private zones

If a resource-critical Internet Telephone is configured for a private zone, and there are not enough resources found within that zone, the search continues to the shared zones within the same customer for an available DSP channel.

However, if a phone is configured in a shared zone, the PBX/Call Server limits its search to the pool of shared DSP channels. That is, the Internet Telephone does not search into the private zones' channels.

When configuring the allocation of shared versus private resources, consideration must be given to the number of private resources that are needed. Enough DSP resources should be configured to prevent the Internet Telephones configured in shared zones from running out of channels.



WARNING

The PBX/Call Server does not search for channels in private zones if it is configured to use only shared zones. Only Internet Telephones configured in the same private zone can use the private zone voice gateway channels.

Since the channels in the private zone are not accessible to Internet Telephones in the shared zone, ensure that only enough private channels are configured to cover the Internet Telephones in private zone. Do not configure more than is needed in the private zone because the shared zone telephones do not have access to these channels.

Interworking with NAT

Network Address Translation (NAT) provides the following benefits:

- the ability to network multiple sites with overlapping private address ranges
- the added security for servers on a private network
- the conservation of public IP address space

A NAT device exists between a private network and a public network. The NAT device maps private addresses to public addresses.

IP Line 3.0 NAT

IP Line 3.0 does not support all the benefits of NAT listed above. IP Line 3.0 implements only a periodic message feature to keep NAT sessions alive when a call is on mute. The periodic messaging prevents an RTP packet stream NAT session from timing out. This could occur when the Internet Telephone is muted and packet transmission is stopped.

To support multiple Internet Telephones behind one NAT device, it is necessary for NAT to map between public and private IP addresses, and ports for each Internet Telephone behind it. The mapping should include both a signaling port and media (voice) port. In a situation where there are multiple NATs between the Internet Telephone and the Voice Gateway Media Card, all NATs on the path have to follow the rules described in the following sections for signaling and media streams.

Mapping is configured and implemented using the NAT device. The IP Line application does not implement any of the mappings.

NAT and Signaling

NAT hides the true identity of the Internet Telephone from the TPS. The TPS knows only the IP address and port number of the Internet Telephone.

Signaling messages between the Voice Gateway Media Card and Internet Telephones are carried by RUDP. Each RUDP connection is distinguished by its IP address and port number. The NAT does one-to-one mapping on the signaling port for each Internet Telephone behind it to support multiple phones. The TPS uses fixed port numbers for signaling. NAT must do a one-to-one mapping for these port numbers. Table 20 on page 78 lists the UDP port number used.

Table 20
Signaling UDP Ports

UDP Port	Device	Use
5000	Internet Telephone	Incoming signaling messages to the Internet Telephone
5100	TPS	Incoming call processing messages to the TPS
4100	TPS	Incoming registration message to Connect Server
7300	TPS	Incoming registration messages to node Master

NAT and Media Streams

The media stream port numbers on the Voice Gateway Media Card use a fixed numbering scheme where the starting number for the port range is configurable. The first port on the card uses the configured starting port number and subsequent ports are numbered in a monotonically increasing manner. Each port has two sequential numbers: one for RTP and one for RTCP.

Note: The NAT has to provide one fixed public port number (5200) for all media streams to and from Internet Telephones behind it.

This port should not be changed at any time, and it should map to port 5200 on the Internet Telephones.

Table 21
IP Line Media Path UDP Ports

UDP Port	Device	Use
5200-5262	Succession Media Card	RTP packets (configurable starting port number - Internet Telephone's port matches it)
5201-5263	Succession Media Card	RTCP packets into Succession Media Card (port number is RTP port number + 1)
5200-5246	ITG-P Line Card	RTP packets (configurable starting port number - Internet Telephone's port matches it)
5201-5247	ITG-P Line Card	RTCP packets into SMC (port number is RTP port number + 1)
5200	Internet Telephone	RTP packets into internet phone (port matches first RTP port of the Voice Gateway Media Card)
5201	Internet Telephone	RTCP packets into Internet Telephone (port matches first RTCP port of the Voice Gateway Media Card)

Mute and Hold Considerations

IP Line 3.0 has to handle two special cases when interworking with NAT. These are mute and hold.

Mute

When a user enables Mute, the TPS sends a Mute Transmit (Tx) command to the Internet Telephone. This forces the telephone to generate silence in the transmit direction. If the telephone is using an evocator that implements silence suppression, for example G.729AB, the telephone sends one silence frame to the far end, and then stops sending any further frames until Mute is cancelled. As a result, data sent from the Internet Telephone stops.

The NAT device sees that the Internet Telephones's UDP connection is not active in the transmit direction and starts aging the translation. Depending on the length of time the call is muted and the duration of the NAT's translation aging timeout value, it is possible for the NAT device to timeout the translation and drop the connection. At this point, all packets coming from the far end would be dropped by the NAT device.

At the time mute is cancelled, the Internet Telephone starts transmitting again. NAT considers this to be a new connection and creates a new translation, sending data to the far end using this new translation. This results in half-duplex voice connection between the Internet Telephone and the far end device. In this case, data sent to the far end device gets there; however, the data coming back is lost.

To solve this problem, the Internet Telephone must periodically send an extra non-RTP packet to the far end to keep the NAT translation alive. This "bogus" packet is sent to ensure that the NAT's session timeout does not expire. The non-RTP packet is constructed to fail any RTP validation tests so it is not played out by the far end device (Internet Telephone or gateway channel).

Hold

The Hold function differs from the Mute function. When an Internet Telephone user places a call on Hold, it closes the audio stream in both the Transmit (Tx) and Receive (Rx) directions. To NAT, it is similar to Mute, and the NAT device begins aging the translation. However, this situation does not require a special treatment on the IP Line application or Internet Telephone firmware. When the call is retrieved from Hold, a new set of open audio stream messages are issued by the TPS and new connections are established. As a result, it does not matter if the NAT device has deleted the earlier session.

Table 22 shows supported operations for Internet Telephones behind a NAT device in IP Line 3.0.

Table 22
IP Line NAT Support Strategy

Requirement	Comment
An Internet Telephone behind a NAT device can be registered.	Yes
An Internet Telephone behind NAT can make a call through the Voice Gateway Media Card channels and also to other Internet Telephones that are not behind the NAT device.	Yes
A call can be established between two Internet Telephones on the same private network (behind the same NAT device).	No
If there are multiple DNs configured on an Internet Telephone, all DNs are able to make and receive calls.	Yes
The user must be able to hold, mute, and retrieve calls.	Yes

Note: While not directly associated with NAT support, the i2002/2004 Internet Telephone firmware continues to be downloaded using Trivial File Transfer Protocol (TFTP). UFTP is not supported. TFTP prevents placing Internet Telephones behind firewalls that have the TFTP port blocked for security reasons. An alternative is to upgrade the Internet Telephones firmware to the latest version prior to placing it behind the firewall.

NAT Configuration

The Element Management Web server and OTM 2.0 GUIs have two new prompts to configure the timer function. A checkbox is used to enable or disable the NAT message. When enabled, a configuration box sets the time (in seconds) between messages sent. The default value is 90 seconds. The configured values apply to all Internet Telephones on the node.

Run-time configuration changes

The ITG Line 2.x applications required the ITG-P Line Card to be disabled and then enabled to activate some administrative changes, and in some cases, a card reboot was required for changes to take effect.

IP Line 3.0 improves this functionality by enabling most changes to be made without disabling or rebooting the Voice Gateway Media Cards. After adding configuration information for a new Voice Gateway Media Card and downloading the BOOTP file to the Leader, a new Voice Gateway Media Card can be added to an existing node without rebooting the other cards.

The following exceptions exist for the changing node properties and these changes are not supported by this enhancement:

- A role change requires a reboot to enable the application to reconfigure itself. That is, changing a Leader to Follower or changing Follower to Leader.
- Changing the node IP, subnet masks, or gateway IP addresses requires a reboot of all cards in the node.
- Changing the IP address of a particular card requires a reboot of that card so it can retrieve its new IP address information.

Therefore, IP Line 3.0 supports only run-time changes for the following:

- changes to the CONFIG.INI file
- add card or delete card changes to the BOOTP.TAB file

CONFIG.INI file

The following are automatically reconfigured after the CONFIG.INI file is downloaded from OTM to the card:

- DSCP bits setting
- configuration of ELAN link to Call Server
- loss plan
- SNMP traps
- Routing Table
- firmware download
- codec selection

The reconfiguration of firmware download is done only in the Master card. New firmware files are retrieved from the new file server. The Master card then sends out a broadcast message to the Follower cards indicating the need for a file update.

BOOTP.TAB file

If a card is added or deleted from the node, the BOOTP server is updated using the new BOOTP.TAB file. The new card can retrieve its parameters with the next BOOTP request.

Element Management on Succession CSE 1000 Rel 2.0 generates the CONFIG.INI and BOOTP files and places them on the Call Server. It then notifies the Voice Gateway Media Cards to retrieve these files. The BOOTP server normally runs on the Signaling Server but also runs on any Voice Gateway Media Card acting as the Master.

OTM 2.0 on the Meridian 1 and Succession CSE 1000 Rel 1.1 system downloads the BOOTP.TAB file to the node's Leader card. The file is then distributed from the Leader to the Follower cards in the node.

Configuration changes have an effect only on new calls. Existing calls are not interrupted. However, there are exceptions:

- If the active Call Server ELAN link's configuration data is changed (for example, a changed IP address), then active calls are released.

If the non-active Call Server is changed (for example, survivable side IP address), then the calls are not affected.

When the ELAN connections are taken down to implement the configuration change, the Internet Telephones and gateway channels registrations are unregistered on the Call Server. The Call Server releases the calls. When the link is re-established, the TPS synchronizes the call states and releases the active calls. Service is interrupted during this re-establishment period and the following are affected:

- new Internet Telephones cannot register
- registered Internet Telephones cannot establish new calls
- the Voice Gateway Media Card's faceplate displays S009

Once the ELAN link comes up, the Line Terminal Proxy Server (LTPS) reregisters the telephones with the Call Server and all service is resumed.

- If the codec list is changed, the Voice Gateway Media Card's DSPs may need to be reloaded. For instance, one DSP image contains G.711, FAX, and G.729A/G.729AB. The other DSP image contains G.711, FAX, and G.723.1. If the user has a node configured with the G.729AB codec and the user performs an administrative change to use G.723.1 (or vice versa), the DSPs must be reloaded.

After the CONFIG.INI file containing the administrative change is downloaded to a Voice Gateway Media Card, the card's DSPs are reloaded as they become idle. For instance, if all DSPs are idle on the card, the new image is loaded to all of them at once. If one or more DSPs have calls active, the DSP is not reloaded until the active calls have released. This can cause some DSPs to be reloaded later than others.

This new functionality is supported by both Element Management and OTM 2.0.

Patching tool

A patch is a piece of code that is inserted or patched into an executable program. The patching tool enables loadware on the ITG-P Line Card and Succession Media Card platforms to be patched or fixed without having to upgrade the card loadware and without service interruption. Patches are stored on the MPLS database.

All patch commands on the ITG-P Line Card, Succession Media Card, and Signaling Server are accessible at the IPL> prompt. These commands are summarized in Table 23 on page 86.

The functionality and syntax is similar to the current Call Server patching tool.

Note: The exception is that the parameter string supplied to the command must be enclosed with double quotes. For example, the syntax for the pload command is pload "patch1.p".

These commands are used to manage patches on the Voice Gateway Media Card. Patches must be downloaded from a workstation to the Voice Gateway Media Card using a modem, an FTP session, or Element Management. Patch files are stored in Flash memory and are loaded into DRAM memory. Once a patch is in DRAM memory it can be activated, deactivated, and its status can be monitored.

The technician must perform the following tasks prior to loading a patch:

- Check that the patch matches the platform's CPU type.
- Check the loadware version on the card.
- Block the installation if there is a mismatch.

The installation of a patch is blocked if either the CPU type or the loadware version of the card is different than the patch. If the installation is blocked, the reason for blocking the install is printed at the CLI. The CPU type and loadware version are also checked during a power-up or reboot cycle. This prevents active patches from being re-installed if the loadware version of the card is changed.

Table 23 on page 86 lists the patch commands.

Table 23
Patch commands (Part 1 of 3)

Command	Description
pload	Loads a patch file from the file system in Flash memory into DRAM memory. The loaded patch is inactive until it is put into service using the pins command. When a patch is successfully loaded, the pload command returns a patch handle number. The patch handle number is used as input to other patch commands (pins, poos, pout, and plis). Syntax: pload "[patch-filename]" where [patch-filename] is the filename or path of the patch file. If a filename alone is provided, the patch must be in the /C:/u/patch directory. Otherwise, the full or relative path can be provided. If the pload command is issued without a parameter, the technician is prompted for the patch filename and other information.
pins	Puts a patch that has been loaded into memory (using the pload command) into service. This command activates a patch. If issued successfully, the pins command indicates that global procedures, functions, or areas of memory are affected by the patch. The technician is then prompted and has the choice to proceed or not to proceed. Syntax: pins "[handle]" where [handle] is the number returned by the pload command If the pins command is issued without a parameter, the technician is prompted to enter a handle.

Table 23
Patch commands (Part 2 of 3)

Command	Description
poos	Deactivates a patch (takes it out-of-service) by restoring the patched procedure to its original state. Syntax: poos "[handle]" If the poos command is issued without a parameter, the technician is prompted to enter a handle.
pout	Removes a patch from DRAM memory. The patch must be taken out-of-service (using the poos command) before it can be removed from the system. Syntax: pout "[handle]" If the pout command is issued without a parameter, the technician is prompted to enter a handle.
pstat	Gives summary status information for one or all loaded patches. For each patch, the following information is displayed: patch handle, filename, reference number, whether the patch is in-service or out-of-service, the reason why the patch is out-of-service (if applicable), and whether the patch is marked for retention or not. Note: Patch retention means that if a reset occurs, then the patch is automatically reloaded into memory and its state (active or inactive) is restored to what it was prior to the system going down. Syntax: pstat "[handle]" If the handle is provided, only the information for the specified patch is displayed. If the pstat is issued without a parameter, information for all the patches is displayed.

Table 23
Patch commands (Part 3 of 3)

Command	Description
plis	Gives detailed patch status information for a loaded patch. Syntax: plis "[handle]" If the pout command is issued without a parameter, the technician is prompted to enter a handle.
pnew	Creates memory patches for the Voice Gateway Media Card. - The release of the patch is assumed to be the same as that of the current load. - The address to be patched is checked to ensure that it is in range. - For each address that is changed, the "old" contents are assumed to be the current contents of that memory address. - If a path is not provided for the new path filename then it is assumed that the patch is in the /C:/u/patch directory. Once a memory patch is created using the pnew command, it is loaded and activated like any other patch. Syntax: pnew Note: The pnew command has no parameter(s).

Patch Directories

There are two patch directories on a Voice Gateway Media Card:

1 /C:/u/patch

This is the default directory for patch files. Patch files should be copied to this directory.

2 /C:/u/patch/reten

The technician uses this directory to store patch retention control files. Do not use this directory to store patches and do not remove files from this directory.

Patch Synchronization Across a Node

The Succession CSE 1000 Rel 2 Element Management Web server provides a mechanism for downloading and putting patches in service across a node.

Patch synchronization across a node cannot be carried out from the IPL> prompt. On Meridian 1 or Succession CSE 100 Rel 1 systems, patch synchronization is a manual operation that must be performed at the CLI of each Voice Gateway Media Card.

Virtual Office

Succession CSE 1000 Rel 2.0 and IP Line 3.0 provides the Virtual Office feature. This feature enables a user to use any Internet Telephone within the network.

The Virtual Office feature provides a call service to "travelling" users who want to use a different physical Internet Telephone (other than the telephone they normally use). Users can login to another Internet Telephone using their DN and pre-configured Station Control Password (SCPW).

Once logged in, users have access to their DNs, autodial numbers, key layout, feature keys, and voice mail indication/access that are configured on their own home-office Internet Telephones. For example, if users go to another office or to a different location within the same office, they can login to any available Internet Telephone and have all the features of their home-office Internet Telephone. When the user logs off the Internet Telephone, the features that were "transferred" to that telephone are removed.

Virtual Office is supported for the i2002 and i2004 Internet Telephones, and the i2050 Software Phone. An i2004 or i2050 user is prevented from accessing this feature from an i2002 Internet Telephone. Table 24 on page 90 shows which user can log in to particular telephones.

Table 24
Virtual Office login from various telephones

Internet Telephone User	Virtual Office login
An i2002 Internet Telephone user...	...can Virtual Office login from i2002, i2004, and i2050.
An i2004 Internet Telephone user...	...can Virtual Office login from i2004 and i2050. ...receives a "Permission Denied" message when the user attempts a Virtual Office login from an i2002 Internet Telephone.
An i2050 Software Phone user...	...can virtually login from i2004 and i2050. ...receives a "Permission Denied" message when the user attempts a Virtual Office login from an i2002 Internet Telephone.

Virtual Office User Allowed (VOUA) and Virtual Office Login Allowed (VOLA) must be configured on the Internet Telephones as follows:

- The Internet Telephone where the user wants to virtually login (destination) must have Virtual Office User Allowed (VOUA) configured.
- The Internet Telephone where the user wants to login from (source) must have Virtual Office Login Allowed (VOLA) configured.

Note: Three failed password attempts to login using the Virtual Office feature locks the user out from Virtual Office login at the Call Server for one hour. The Call Server lock can be removed by an administrator using an LD 32 command to disable and re-enable that TN. Refer to *Maintenance* (553-3023-510) for more information.

An Internet Telephone registers using the TN (in its EEPROM), and then a valid user id and password are used to determine the Home TPS for the Internet Telephone during the Virtual Office connection. A Succession CSE 1000 Gatekeeper is required if the Home TPS is not the TPS where the Internet Telephone is registered when the Virtual Office login is initiated.

Virtual Offices provides the following capabilities:

- 1 A network-wide connection server (Gatekeeper) is equipped to provide addressing information of call servers, based on a user's DN.
- 2 The user can enter a key sequence at an Internet Telephone to initiate the login sequence. The user enters their current network DN and a user-level password. The password is the Station Control Password configured in LD 11. If a SCPW is not configured, the virtual office feature is blocked.
- 3 A user logs out when leaving the location.

For more detailed information about Virtual Office, see *Internet Terminals Description* (553-3001-217).

Branch Office

The Succession Communication Server for Enterprise (CSE) 1000 Branch Office provides a means of extending Succession CSE 1000 features to one or more remotely-located Branch Offices.

The Succession CSE 1000 Branch Office is a feature set of the equipment and software that a secondary location needs to centralize the call processing of its IP-based communications network. The Call Server at the Main Office provides the call processing for the Internet Telephones in both the Main Office and Branch Offices. The H.323 WAN Gateway in the Branch Office provides access to the local Public Switched Telephone Network (PSTN).

The Succession CSE 1000 Branch Office is connected to the Main Office over a virtual trunk on a WAN. Internet Telephone calls and IP network connections are controlled by, and come from, the Main Office. If the Main Office fails to function, or if there is a network outage, the Succession System Controller (SSC) at the Branch Office provides service to the telephones located in the Branch. The telephones then survive an outage between the Branch Office and the Main Office.

The basic hardware of a Branch Office includes the H.323 WAN Gateway and the Signaling Server. The H.323 WAN Gateway provides access to the local PSTN for users in the Branch. It also provides support for analog devices such as fax machines or telephones in the Branch Office.

For detailed information about Branch Office, refer to *Branch Office* (553-3023-221).

802.1Q Support

Firmware version 1.3x of the i2002/2004 Internet Telephone enables 802.1Q support. 802.1Q support enables the definition of virtual LANs (VLANs) within a single LAN. This improves bandwidth management and limits the impact of broadcast and multicast messages.

Supporting 802.1Q on the Internet Telephones enables VLAN configuration and packet prioritization to be simplified. These are configured at the Internet Telephone instead of at the network switch port for each telephone. The "p" bits within the 802.1Q standard enable packet prioritization at Layer 2 and this improves network throughput for VoIP data. The network switching equipment must still be capable of recognizing and processing the 802.1Q header. However, 802.1Q support reduces the configuration necessary to add VLAN and prioritization functionality.

Only the i2002 and i2004 Internet Telephones generate the 802.1Q header. The ITG-P Line Cards and Succession Media Cards do not. As a result, the switch ports for the Voice Gateway Media Card TLAN ports must be configured as untagged ports so the header is removed.

VLAN ID and Priority Field

The 802.1Q VLAN ID:

- Provides a higher level of security between segments of internal networks.
- Breaks large networks into smaller parts to prevent consumption of bandwidth to broadcast and multicast traffic.

The 802.1Q Priority field enables classification of UDP traffic for prioritization through network Layer 2 switching.

802.1Q Tag

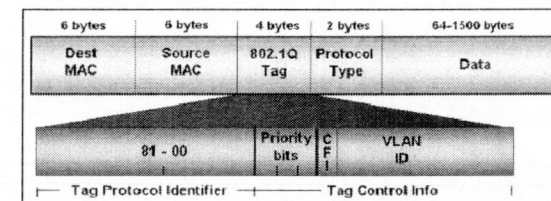
A standard Ethernet frame contains a header consisting of a 6-byte destination MAC address, a 6-byte source MAC address, and a 2-byte protocol identifier. Following the header is a data area.

The 802.1Q standard specifies a new format of Ethernet frame. The 802.1Q protocol standard makes extensions to the Ethernet frame by adding four additional bytes to the Ethernet header. One of the four 802.1Q extension fields applies to QoS. This field is a 3-bit priority field that is referred to as 802.1p. These priority bits are shown in Figure 5 on page 94.

The 802.1Q formatted frame is identical to a standard Ethernet frame, with the exception of the 4-byte 802.1Q tag that is inserted between the source MAC address and the protocol identifier. The first 16 bits of the 802.1Q tag field is the Tag Protocol Identifier containing 8100 (hex) (the 802.3 tag format), enabling the Ethernet interface to distinguish it from standard Ethernet frames. The last 16 bits of the 802.1Q tag contain the following information:

- a 3-bit Priority field (the 802.1p bits)
This field represents the user priority level (7 is the highest).
- a 1-bit Canonical Field Identifier (CFI)
This field is always set to 0.
- a 12-bit VLAN ID field

Figure 5
802.1Q Tag Format



Automatic VLAN ID Configuration

As part of the 802.1Q feature, an option to automatically discover the VLAN ID using DHCP exists. This process greatly reduces the configuration steps since manually entering the VLAN ID data is not required.

When automatic VLAN discovery using DHCP is used to obtain the VLAN ID and the Internet Telephone is configured using DHCP, the following occurs automatically:

- The i2002 and i2004 Internet Telephones perform an initial DHCP Discovery Request in the default VLAN.
- The DHCP server returns a DHCP ACK message with an IP address in the data VLAN and one or more voice VLAN IDs in the vendor-specific field (refer to Appendix G on page 719 for an example of the DHCP configuration strings).
- The Internet Telephone reads the voice VLAN ID(s) and saves them.
- The Internet Telephone rejects the DHCP offer (accepts it but immediately gives up the lease).
- The Internet Telephone reboots and sends a DHCP Discovery Request with the first VLAN ID from the saved list. This is repeated for each VLAN ID in the list until a response is received.

This works because the Layer 2 switch discards every DHCP Discovery Request it receives from the Internet Telephone if the VLAN ID does not match the VLAN IDs configured on the port. When the Internet Telephone sends a DHCP Discovery Request with the port's configured VLAN ID, the packet passes into the network and the DHCP server's ACK message is passed back.

- When a DHCP ACK message is received, the Internet Telephone accepts the offer and saves the IP address, Node IP address, and other IP parameters.

The Internet Telephone's use of 802.1Q

The i2002/i2004 Internet Telephone supports 802.1Q as follows:

- 802.1Q can be enabled or disabled at boot time using manual configuration or control downloaded from the TPS.
 - If 802.1Q is disabled, standard Ethernet frames are transmitted.
 - If 802.1Q is enabled, all frames transmitted by the Ethernet driver have the 802.1Q tag bytes inserted between the source MAC address and the protocol type field. The tag protocol identifier field contains 8100 (hex) and the CFI bit set to 0.
- By default, when 802.1Q is enabled, the priority bits of all frames are set to 6 (octal) and the VLAN ID is set to 000 (hex). The GUI and TPS configured values override these values.
- The Internet Telephone's Ethernet driver receives any Ethernet frame destined for it, regardless of whether 802.1Q is enabled or whether the received frame is an 802.1Q tagged frame. The only exception is any 802.1Q tagged frame with the CFI=1. In this case, the frame is discarded.
- The Internet Telephone's Ethernet driver strips the 802.1Q tag information from the frame prior to passing it on to the IP stack. Priority and VLAN information on received frames are not preserved and are ignored.

Configuration of the Internet Telephone's 802.1Q

The 802.1Q support for the Internet Telephones is configured and controlled using the telephone's user interface or DHCP. The DHCP approach eliminates the need to manually set the VLAN ID during the installation. To configure 802.1Q, set the following: "p" bits and VLAN ID.

Setting the "p" bits

By default, the 3-bit field "p" bits are set to 110b (6), which is the value recommended by Nortel Networks. The "p" bit value can be changed using either OTM or Element Management. The same "p" bit values are assigned to the RTP media stream and the UNISim control signaling stream. The TPS sends the "p" bit value to the telephone using a UNISim messaging.

There are two fields in the OTM and Element Management GUI used to set the "p" bits:

- 1 A **checkbox** that, when checked, means the priority bits should be set to the value specified by the 802.1Q priority bit value field. If the checkbox is unchecked, the i2002/i2004 Internet Telephone sends out the default priority of 6.
- 2 A **802.1Q priority bit value field**. This field sets the value that the i2002/i2004 Internet Telephones sends out. The range is 0-7.

Setting the VLAN ID

The contents of the VLAN ID field can be specified on a "per interface" basis. There is only one network interface on the i2002 and i2004 Internet Telephones, therefore, the setting of the VLAN ID field is a "global" setting. That is, all packets transmitted by the Internet Telephone have the same VLAN ID.

The VLAN ID is specified as follows:

- the default VLAN ID is 000 (hex)
- the VLAN ID can be set during a manual configuration of the Internet Telephone using the telephone keypad, or automatically retrieved using DHCP (automatic VLAN discovery).

Note: For more information about manual or automatic Internet Telephone configuration, refer to *Internet Terminals Description* (553-3001-217).

Some implementation requirements of the Automatic VLAN Discovery using DHCP are:

- 1 A DHCP server IP address pool must exist for each subnet (also VLANs). This is standard DHCP operation. The requirement would be the same for PCs or Internet Telephones.
- 2 A DHCP server should not exist in more than one VLAN at one time (one subnet for each VLAN), unless the link to the DHCP server is tagged and the DHCP server can recognize this. With an untagged link to the DHCP server, traffic could originate on one VLAN and end up on the other VLAN. In this case, the VLAN using DHCP feature does not work.

- 3 Voice and data subnets must be separate if the three-port switch with VLANs is being used.
- 4 A layer three switch (or router) with a Relay agent must be used because traffic from the voice VLAN to the data VLAN must be routed. Presumably, the DHCP server is on the data VLAN. Without a Relay agent a DHCP server must exist on each subnet.
- 5 At least two IP address pools are used on the DHCP server. One for the Voice VLAN/subnet and another for the Data VLAN/subnet. Additional pools can be added as required as long as one IP address pool per subnet and VLAN is used. A relay agent is required if it is a PC-only network.

Control of the Internet Telephone's 802.1Q

The 802.1Q header in the outgoing packets from the i2002 and i2004 Internet Telephones is enabled by one of the following:

- If the Internet Telephone's VLAN GUI response is set to 1, then the 802.1Q functionality is enabled. All packets from the Internet Telephone have the 802.1Q header as part of the Ethernet frame.
- If the Internet Telephone's VLAN GUI response is set to 2, then the 802.1Q functionality is enabled after the DHCP response is received with the VLAN ID.
- If the OTM/Web server configuration enables the use of the "p" bits, once downloaded to the Internet Telephone, the 802.1Q functionality is enabled.

Table 25 on page 99 shows the relationship between the data configured at the TPS and the Internet Telephone (using the GUI or DHCP discovery) and the resulting 802.1Q data sent.

Table 25
Relationship between TPS and GUI configuration to 802.1Q

		Phone 802.1Q setting (by GUI or DHCP)	
		Disabled	Enabled VLAN ID = nnn
OTM / Element Management 802.1p Setting	Disabled	Normal Internet Telephone Frames	802.1Q Tagged Frames Priority = 6 CFI = 0 VLAN ID = nnn
	Enabled; Priority = P	802.1p Priority Tagged Frames Priority = P CFI = 0 VLAN ID = 000	802.1Q Tagged Frames Priority = P CFI = 0 VLAN ID = nnn

802.1Q and the Voice Gateway Media Cards

The ITG-P Line Card and Succession Media Card cannot send the 802.1Q header because the cards' operating system does not support it. The switch ports connected to the ITG-P Line Card's or Succession Media Card's TLAN should be configured for untagged operation so if a 802.1Q header is present it is stripped before a packet is passed to the card.

The configuration in OTM and Element Management is for the control of the priority bits in the 802.1Q header sent by the Internet Telephones only.

Data Path Capture tool

IP Line 3.0 contains the Data Path Capture tool. A built-in utility used to capture audio information. This tool can help debug audio-related gateway problems and allows after-the-fact analysis of what the user heard.

The audio data captured using the Data Path Capture tool is saved in DRAM in either a circular or a fixed-duration buffer.

- A circular buffer functions in such a way that when the end of the buffer is reached, the new audio data overwrites data at the beginning of the buffer. As a result, a circular buffer always has the most current data contained in the buffer.
- A fixed-duration buffer captures data until the buffer is full. When the buffer is full, the capture process stops.

Once the data is captured, it can be saved to a device on the LAN (using FTP) or it can be written to the flash memory card in the PC Card slot on the Voice Gateway Media Card.

The Data Path Capture process is controlled by a set of CLI commands.

i2002 and i2004 Internet Telephone Firmware

New Firmware, version 1.38 minimum

Version 1.38 of the i2004 and i2002 Internet Telephone firmware (F/W) is the minimum version that is supported with IP Line 3.0. This firmware version contains the following changes:

- upgrades the pSOS+ operating system from a special version to the normal release version.
- enables the support of Virtual Office / Branch Office
- the new operating system version enables the support of 802.1Q (see "802.1Q Support" on page 93).

Firmware Download

In ITG Line 2.0-2.2, the i2004's firmware file was downloaded from OTM to each of the Voice Gateway Media Cards and saved in a directory on the card's Flash disk. As each i2004 Internet Telephone registered, the TPS determined if a firmware upgrade was required and directed the i2004 Internet Telephone (requiring the upgrade) to the FTP server of the card.

The introduction of the i2002 Internet Telephone in IP Line 3.0 requires changes to the way the firmware file is handled.

When the firmware file is downloaded from OTM, it is compressed as it is stored on the /C: drive. File compression reduces the firmware file to less than 900 K. However, the /C: drive Flash disk space is limited on the ITG-P Line Card.

The Internet Telephones may not be distributed with the correct version of the F/W file pre-loaded except in cases where the TFTP download to the Internet Telephone is blocked. For example, if the Internet Telephone is located behind a firewall.

The Internet Telephone normally does not have to be pre-loaded with the firmware file because, during normal operation, the Internet Telephone's firmware is automatically upgraded as part of the registration to the TPS. If the firmware cannot be upgraded because of firewall restrictions, then the Internet Telephone must be upgraded with the current firmware version before distributing the telephone.

Firmware filenames

The Internet Telephone firmware files are released on CD-ROM. The files are also available from the Nortel Networks Web site (see Appendix G: "Downloading IP Line files from Nortel Networks Web Site" on page 719).

The Internet Telephone firmware files are labelled as follows:

- 0602Bnn.BIN is the filename for the i2004 Internet Telephone firmware where Bnn = F/W version 1.nn.
- 0603Bnn.BIN is the filename for the i2002 Internet Telephone firmware where Bnn = F/W version 1.nn.

If the external file server option is used (in OTM or Element Management) for firmware distribution with a node, the files must be renamed before being placed on the server:

- 0602Bnn.BIN must be renamed to i2004.fw
- 0603Bnn.BIN must be renamed to i2002.fw

For the external file server options:

- see Procedure 22 on page 297 for OTM
- see Procedure 39 on page 359 for Element Management

Meridian 1 and Succession CSE 1000 Rel 1**Default location of firmware files**

The firmware file for the i2004 Internet Telephone is stored in the C:/FW directory. This is the same directory that is used in the IP Line2.0-2.2 products. The firmware file is downloaded and saved to this directory when the user checks the firmware download checkbox in the OTM Synchronize/Transmit dialog and presses the Transmit button. The IP Line application saves the file i2004.fw as fwfile.1 or fwfile.2 for backwards compatibility. The filename is chosen such that the oldest of the two files on the card is overwritten. Then at card bootup time, if the firmware file is not retrieved from the external server or the /A: drive, the /C:/FW directory is accessed and either the fwfile.1 or fwfile.2 is uncompressed. The newer of the fwfile.1 or the fwfile.2 is uncompressed.

Firmware file management with IP Line 3.0

The IP Line 3.0 operates the same as the IP Line2.0-2.2 product. The firmware file is stored and retrieved from the local /C:/FW directory.

The IP Line 3.0 application searches for the firmware first at the file server, then in the /A:/FW directory, and finally in the /C:/FW directory.

- Normally the file server is not configured in OTM 2.0. OTM 2.0 places IP address 0.0.0.0 in the CONFIG.INI file for the file server address. If an address of "0.0.0.0" (the default) is read from the file, the IP Line 3.0 application ignores the file server settings. As a result, the normal search ends with the firmware file being retrieved from the /C:/FW directory.
- If a file server address is configured, the IP Line 3.0 application operates as described for the Succession CSE 1000 Rel 2.0 system. The file is downloaded into the /ums directory in memory. In order for all of the cards to get the same firmware files, the technician must ensure that the configured file server is up and running before any of the cards boot up.

The "/A:" drive (faceplate PC Card slot) of the Voice Gateway Media Card can also be used with a PC Card containing the firmware files. The card is specified as the server and the file directory specifies the "/A:/FW" drive.

Download Protocol

The TFTP download mechanism is used in IP Line 3.0. The Master card notifies the Followers about changes to the status of the firmware file using a broadcast on the TLAN interface.

Note: UFTP is not supported.

Bootup Scenarios

If the Master is unable to retrieve a firmware file, the upgrade policy is set as "Never". When the upgrade policy is set to "Never", the Internet Telephone's firmware version is not checked and the telephone registers with the firmware version that is currently on the telephone.

If the Master card reboots, the Election process selects another Voice Gateway Media Card as the Master. That Voice Gateway Media Card has all firmware files in its memory. When the original Master card finishes rebooting, it becomes the Master and does the normal Master startup procedure for retrieving the firmware files.

In a power-on situation, where all cards reboot together, the first card that is elected Master retrieves the firmware files from the server.

Succession CSE 1000 Rel 2.0**Default location of firmware files**

For Succession CSE 1000 Release 2.0 Normal configuration, the default storage location for the firmware files is on the Signaling Server in the /u/fw directory. The firmware file is downloaded to this directory when the user selects the file in Element Management and presses the Transmit button.

Firmware file management with IP Line 3.0

Due to the limited flash drive space on the Voice Gateway Media Cards, a new mechanism for managing the Internet Telephone firmware files is required. IP Line 3.0 manages the firmware file in the following manner:

- 1 There is one firmware file for each telephone type. These files are saved and retrieved in one of the following two locations:
 - a to/from a file server
(The file server can be a dedicated external server, the Call Server, or a Voice Gateway Media Card.)
 - b to/from a Master card's RAM device
- 2 The server's information is configured in Element Management and the information is saved in the CONFIG.INI file. The server's IP address, routing table, file path, user name, and password are specified during configuration time.
- 3 When the Master card boots, it searches for the firmware files on the specified server.
 - a If found, they are retrieved and stored on the RAM drive in the /ums directory.
 - b Otherwise,
 - i. for a Voice Gateway Media Card, the Master card continues to search for the firmware files in the local A:/fw directory and then the C:/fw directory until the files are found.
 - ii. for a Signaling Server, the Master card attempts to search for the firmware files in the /u/fw directory, and then the /A:/fw directory.

The firmware retrieval process is much faster using IP Line 3.0, as the file is written to the RAM drive. ITG Line 2.0-2.2 required more time because the files are written to the Flash drive.

- 4 When a Follower card boots, it looks for the firmware files on the Master card's RAM drive in the /ums directory.

If the Master has not yet retrieved the files, the Follower waits until the Master sends notification that the firmware files are retrieved. Using FTP, the Follower transfers the files from the Master and stores them in the /ums directory on its RAM drive.
- 5 Once a firmware file is found and stored in the card's RAM drive, the upgrade manager parses the file and updates its policy based on the firmware version it received from file.
- 6 Both the i2002 and i2004 Internet Telephones are checked against the upgrade policy at the time they register. If a firmware update is required, the firmware is downloaded from the Signaling Server or the Voice Gateway Media Card's TFTP server to the Internet Telephone.

The firmware file for the i2004 Internet Telephone is saved as i2004.fw and is saved as i2002.fw for the i2002 Internet Telephone. These filenames are required for the upgrade manager to find certain files in either the standalone file server or the Master card's RAM drive.

In order for all Voice Gateway Media Cards to obtain the same firmware files, the technician must ensure that the configured file server is running before any of the Voice Gateway Media Cards boot up.

In Succession CSE 1000 Release 2.0, the Signaling Server acts as the file server and the Master function is on the Signaling Server. As a result, the time to download the firmware files from the file server to the Master is eliminated.

The /A: drive (the PC Card slot on the card's faceplate) of the Voice Gateway Media Card can also be used with a PC Card containing the firmware files; the Voice Gateway Media Card is specified as the server and the file directory specifies the /A: drive.

Graceful Disable

The DISI command in LD 32 can be used to disable the Voice Gateway Media Card's gateway channels when they become idle. This command removes gateway call traffic from a Voice Gateway Media Card, however, it does not remove the Internet Telephones registered to the Voice Gateway Media Card. Therefore, even after the gateway channels are disabled, all telephones registered to the card are impacted when the card is unplugged or reset. Also, if a Voice Gateway Media Card or Signaling Server is the node Master when it is removed, the Internet Telephone registration service is interrupted until the next election occurs.

This Graceful TPS enhancement provides a card level CLI command that disables the LTPS service on the Voice Gateway Media Card or Signaling Server.

The Graceful TPS command:

- prevents new Internet Telephones from registering
- soft resets any idle, registered Internet Telephones

Since the LTPS does not accept new registrations, the Internet Telephones register with another card's LTPS after the reset. Eventually, all Internet Telephones are registered with other LTPSs and the card can be removed without impact to any users.

In addition, if the "to-be-out-of-service" Voice Gateway Media Card is the node Master, an election is held immediately when the LTPS service is disabled to transfer the mastership to another card. On the Signaling Server this is not done, as it would interfere with the Virtual Trunk's redundancy feature that depends on the node mastership. The LTPS is still disabled on the Signaling Server so all Internet Telephones are moved to another TPS. However, if the Signaling Server is the node Master, the mastership remains.

Operation of the TPS DISI

The Graceful TPS Disable is controlled from the CLI of the card. When the disiTPS command is executed on the card's TPS, the following occurs:

- The card does not accept any new registration requests.
- The card soft resets all registered Internet Telephones that are in the idle state and redirects the telephones to the node Master.
- The card soft resets the remaining busy registered Internet Telephones after they release their active call.
- If the card is node Master, an election is held to transfer the mastership. This occurs only on the Voice Gateway Media Card. The Signaling Server's node mastership is not transferred.

Feature Operation of the Voice Gateway DISI

The Voice Gateway can also be disabled from the CLI of a Voice Gateway Media Card. When the disiVGW command is executed, the following happens on that card's Voice Gateway:

- Idle gateway channels are unregistered.
- A busy gateway channel is unregistered when it becomes idle.

Note: Care should be taken with this command to avoid a problem that can occur when calls are placed on hold. When a telephone has a call on hold, the voice gateway channel on the card is idle; however, it is still reserved in the Call Server. If the Voice Gateway is still disabled when the call is taken off hold, the call does not have a speech path. It is recommended that the LD 32 DISI command is used for disabling the gateway channels.

Graceful TPS CLI Commands

The following Graceful TPS CLI commands are added to the IP Line shell:

Table 26
Graceful TPS commands (Part 1 of 2)

Command	Description
disiTPS	Disables the LTPS service only. No new Internet Telephones are registered on the card and all registered telephones are reset when they become idle. This command applies to both the Voice Gateway Media Card and Signaling Server. On the Signaling Server, this command affects only the LTPS. It does not affect the virtual trunks or gatekeeper components, which means the node mastership is not moved to another TPS.
disiVGW	Disables the Voice Gateway only. All Voice Gateways unregister with the Call Server when they become idle. This command is applicable only to the Voice Gateway Media Card or the standalone IP Line application.
disiAll	Disables both the LTPS service and the Voice Gateway channels. This command is a combination of both disiTPS and disiVGW commands. On the Signaling Server, this command affects only the LTPS. It does not affect the virtual trunks or gatekeeper components, which means the node mastership is not moved to another TPS.

Table 26
Graceful TPS commands (Part 2 of 2)

Command	Description
enaTPS	Enables the LTPS service. This command is used after the disTPS command to bring the LTPS back into service. This command applies to both Voice Gateway Media Cards and the Signaling Server. On the Signaling Server, this command affects the LTPS only. It does not affect the virtual trunks or gatekeeper components.
enaVGW	Enables the Voice Gateway. All gateway channels register with the Call Server. This command is applicable only to the Voice Gateway Media Card or the standalone IP Line application.
enaAll	Enables both the LTPS service and the Voice Gateway channels. This command is a combination of both enaTPS and enaVGW commands. On the Signaling Server, this command affects only the LTPS. It does not affect the virtual trunks or gatekeeper components.

Maintenance Audit enhancement

The ITG Line 2.2 product on the ITG-P Line Card introduced a background audit that watched for tasks that go into a suspended state. Under normal operation, a task should not go into suspended state. However, if it occurs, the card's processing is affected.

ITG Line 2.2 Operation

With the ITG Line 2.2 application, if the audit task finds a suspended task, it performs the following:

- outputs a stack and register dump to the debug port
- outputs a file on the /C: drive
- resets the card

This function provides an automatic way to return the card to service and provides critical debug information. The information is output to the EXCPLOG.n files (where n is a number from 0-3) that are located in the /C:/LOG directory. The new information is placed in these files where it cannot be overwritten by the usual information output to the SYSLOG file when the card reboots.

The auditRebootSet command was the single maintenance audit CLI command that was provided with ITG Line 2.2. The auditRebootSet command disabled the card reboot if any task was found in a suspended state.

IP Line 3.0 Enhancement

This audit feature is enhanced in IP Line 3.0. This new enhancement differentiates between tasks that are critical and non-critical.

- A critical task is any task that the IP Line application needs to function. When a critical task is not functioning properly, it causes noticeable degradation in the IP Line application.
- A non-critical task is any other task that does not cause noticeable degradation to the IP Line application.

If a critical task is found suspended, the stack and register information is dumped and the card is then reset. If a task on the critical task list disappears, it is treated as a suspended task. Therefore, a missing critical task triggers a reboot and a missing non-critical task does not trigger a reboot.

If a non-critical task is found suspended, the information is dumped but the card is not reset. The card is reset when the Voice Gateway Media Card clock reaches 2:00 AM (default reset time). The reset time is configurable from the CLI. This eliminates card resets that impact service for non-critical tasks by delaying them to a non-service impacting time.

Additional CLI commands have been added enabling any task to be marked as critical or non-critical, regardless of its default designation. This could be used, for example, to mark a "misbehaving" task as non-critical to avoid a card reset. This would enable the problem to be debugged.

The Maintenance Task Audit is available only for the IP Line 3.0 application running on the ITG-P Line Card and Succession Media Card. It is not available on the Signaling Server as it does not have the exception handler, stack dump, and syslog file functions of the other cards.

Critical Task List

All application tasks default to the critical task list. These applications include: TPS, VTM, SET, VTILIB, UMS, UMC, RDP, VGW, RTP, RTCP, ELC, baseMMintTask, and A07.

The following VxWorks system tasks are also on the critical task list: tShell, tNetTask, tExcTask, and tTelnetd.

All other tasks are on the non-critical task list. The monitor task is called tMonTask.

Any data entered at the CLI that deviates the operation from the default is saved in the /C:/CONFIG/AUDIT.INI text file. The contents of the file are loaded as the application boots up and provides the required non-volatile storage for entered settings. It is applicable only to the card on which it resides. It can be manually copied from one card to all other cards in the node if desired.

History File

A history file is created when the card starts. The text file is called audit.his and it is stored in the /C:/LOG directory. This file contains a list of the problems found and the actions taken by the maintenance audit. The audit.his file has a fixed size of 4096 bytes.

The most recent records in the file overwrite the oldest records with newer events appear at the beginning of the file. A record in the file is a one-line string with maximum size of 256 characters.

The format for the records in the history file is:

index : (timeString) TMxx taskName: DescriptionString

where:

- index – monotonically increasing record count; wraps after 9999 events
- (timeString) – the time the event was detected
- TMxx – record type: 0-reboot, 1-Suspend, and 2-TaskDisappear
- taskName – the name of problematic task
- DescriptionString – a description of the action taken

An example of the output follows.

IPL> auditHistoryShow

0001 :(APR 25 12:26:25) TM01 tCSV:Suspend

0002 :(APR 25 12:26:50) TM01 tSET:Suspend

0003 :(APR 25 12:26:50) TM00 tExcTask:Reboot

0004 :(APR 25 12:35:55) TM02 tELC:Disappear

0005 :(APR 25 12:35:55) TM00 tELC:Reboot

0006 :(APR 25 12:48:27) TM01 tUMC:Suspend

0007 :(APR 25 12:48:27) TM00 tExcTask:Reboot

0008 :(APR 25 13:15:56) TM01 tUMC:Suspend

0009 :(APR 25 13:15:56) TM00 tExcTask:Reboot

0010 :(APR 25 13:29:35) TM01 tLogTask:Suspend

0011 :(APR 25 13:45:35) TM01 tLogTask:Suspend

The Maintenance Audit CLI commands

There are five CLI commands that support the maintenance audit function as outlined in Table 27.

Table 27
Maintenance Audit commands (Part 1 of 2)

Command	Description
auditShow	Displays the following information: - whether a card reboot is enabled - the time a card reboot will occur if a non-critical task is found suspended - a list of all tasks being monitored and their designation (critical or non-critical) Example: <pre>IPL> auditShow</pre> Reboot when detect a suspended task --- Disabled Critical Task: tTPS tVTM tSET tVTI tUMS tUMC tRDP tPBX tVGW tRTP tRTCP tELC baseMMintTask tA07 tShell tNetTask tExcTask tTelnetd Non-Critical Task: tTest
auditHistoryShow	Displays the contents of the audit.his file.
auditRebootSet 0/1	Globally disables the card reboot from this audit task. By default, this is set to 1. If it is set to 0, no card reboot occurs when a suspended task is found for critical or non-critical tasks. The debug information is dumped; however, recovery requires a manual reset of the card.

Table 27
Maintenance Audit commands (Part 2 of 2)

Command	Description
auditRebootTimeSet "timeString"	Sets the reset time for non-critical tasks to the value defined by the timeString parameter. The timeString is formatted as HH:MM and is in 24-hour clock format. By default, the time is set to 02:00 (2 AM).
auditTaskSet tTaskName, 0/1	Forces a task to be considered critical or non-critical. This command overrides the audit's default setting for the task. The tTaskName parameter specifies the task (the VxWorks taskname), as displayed by the "I" command. The value of 0 marks the task as non-critical, the value of 1 marks it as critical.

Hardware Watchdog Timer

A hardware watchdog timer is enabled on the ITG-P Line Card and Succession Media Card. This functionality adds further robustness to the existing exception handler and maintenance task audits.

The hardware watchdog timer handles scenarios such as the following:

- the CPU failing
- the code running and not triggering an exception
- resetting the card and bringing it back to normal operation

The timer runs on the ITG-P Line Card and Succession Media Card processors. The card's main processor is polled every 20 seconds. If three pollings are missed, then the card is reset. This gives the main processor 60 seconds to respond, covering most normal operating conditions.

A reset reason is saved when a card resets. The reset reason is displayed as a message during the startup sequence and appears in the SYSLOG file.

The following are examples of reset reasons:

- JAN 04 12:17:45 tXA: Info Last Reset Reason: Reboot command issued
Output after card reset using the CLI command cardReboot.
- JAN 04 12:17:45 tXA: Info Last Reset Reason: Watchdog Timer
Expired
Output after card reset due to watchdog timer expiration.
- JAN 04 12:17:45 tXA: Info Last Reset Reason: Manual reset
Output after card reset due to either the faceplate reset button press or a power cycle to the card.
- JAN 04 12:17:45 tXA: Info Last Reset Reason: Unknown
Output after card reset due either the card F/W not supporting the reset reason or a corruption of the reset reason code.

The last reset reason can also be displayed at any time by entering the lastResetReason CLI command.

The Watchdog Timer and the Voice Gateway Media Card firmware

The application starts the Watchdog Timer as part of the application startup process. The timer is started only if the application's check of the firmware version indicates the card's firmware supports the Watchdog Timer function.

Required Voice Gateway Media Card firmware version

A firmware upgrade can be required on the Voice Gateway Media Cards to invoke the Watchdog Timer functionality:

- ITG-P Line Card – Version 5.3 of the firmware file is the required minimum to enable the Watchdog Timer functionality on the ITG-P Line Card. To upgrade the firmware version of the ITG-P Line Cards to support the Watchdog functionality, see Procedure 90 on page 590.
- Succession Media Card—Version 6.0 of the firmware file is the required minimum to enable the Watchdog Timer functionality on the Succession Media Cards. To upgrade the firmware version of the Succession Media Card to support the Watchdog functionality, see Procedure 91 on page 592.

Codecs

Codec refers to the voice coding and compression algorithm used by the DSPs on the Voice Gateway Media Card. Different codecs provide different levels of voice quality and compression properties. The specific codecs and the order in which they are used, are configured in the TPS and Meridian 1 and Succession CSE 1000.

Table 28 on page 118 shows which codecs are supported on the Meridian 1 and Succession CSE 1000 Rel 1.1 systems, and on the Succession CSE 1000 Rel 2.0 systems:

Table 28
Supported codecs

Codec	Supported on Meridian 1 and Succession CSE 1000 Rel 1.1	Supported on Succession CSE 1000 Rel 2.0
G.711	Yes	Yes
G.729A	Yes	Yes
G.729AB	Yes	Yes
G.723.1	No	Yes
T.38	No	Yes
G.711 Clear Channel	No	Yes

Note 1: The G.723.1 codec is supported only on Succession CSE 1000 Release 2.0 system. The supported G.723.1 codec has bit rates of 5.3 Kbps and 6.3 Kbps. The user can configure the G.723.1 codec with a 5.3 Kbps bit rate; however, the system accepts both G.723.1 5.3Kbps and 6.4Kbps from the far end.

Note 2: The T.38 and G.711 Clear Channel codecs are supported for fax calls and are supported only on the Succession CSE 1000 Release 2.0 system. T.38 is the preferred codec type for fax calls over virtual trunks. However, the G.711 Clear Channel codec is used if the far end does not support the T.38 codec.

For detailed information about codecs, refer to "Codecs" on page 140.

Capacity Engineering Guidelines

Contents

This section contains information on the following topics:

Reference list	119
Overview	120
Capacity engineering	120
IP Line capacity	121
Capacity engineering considerations	124
Traffic capacity of Voice Gateway Media Cards when supporting Internet Telephones	125
ISM parameters	127
Internet Telephone Engineering	128
Traffic and Service Circuits	128
Gateway Channels Traffic Engineering	129
Real time factors	130
Product compatibility with other IP products	131

Reference list

The following are references for this section:

- *Capacity Engineering* (553-3001-149)
- *Features and Services* (553-3001-306)
- *Planning and Engineering Guidelines* (553-3023-102)

Overview

This chapter provides capacity engineering guidelines to help plan and engineer the Meridian 1 and Succession Communication Server for Enterprise 1000 to support the ITG-P Line Card and the Succession Media Card, as well as the i2002 and i2004 Internet Telephones, and the i2050 Software Phone.

Refer to "IP Network Engineering Guidelines" on page 133 for IP Network Engineering information.

Refer to "Configuration of the DHCP Server" on page 211 for engineering guidelines to set and configure the Dynamic Host Configuration Protocol (DHCP) server to support the Voice Gateway Media Card and Internet Telephones.

Capacity engineering

This section explains how to calculate Meridian 1 and Succession CSE 1000 system capacity when engineering the ITG-P Line Card and Succession Media Card for an Internet Telephone.

IP Line capacity

Table 29 lists the system IP Line capacity for cards, telephones, and gateway ports.

Table 29
System IP Line capacity (Part 1 of 3)

Parameter	Capacity
ITG-P Line Cards in each system	Each card requires two slots (subject to EMC restriction, see "Electro-magnetic compatibility (EMC)" on page 672)
— Option 11C or 11C-Mini	— Option 11C - 2 cards in each cabinet (Class B rating) — Option 11C-Mini - 2 cards in each cabinet combination (Main + Expansions) (Class B rating)
— Option 51C, 61C, 81C, and 81C CPPII	— 4 cards in each IPE cabinet (Class B rating)
— Succession CSE 1000 Release 1.1	— 2 cards in each Media Gateway and 2 cards in each Media Gateway Expansion (Class A rating)
— Succession CSE 1000 Release 2.0	— 2 cards in each Media Gateway and 2 cards in each Media Gateway Expansion (Class A rating)

Table 29
System IP Line capacity (Part 2 of 3)

Parameter	Capacity
Succession Media Cards in each system	Each card requires one slot (subject to EMC restriction, see "Electro-magnetic compatibility (EMC)" on page 672)
— Option 11C or 11C-Mini	— No limit (Class B rating)
— Option 51C, 61C, 81C, and 81C CPPII	— Maximum of 10 cards for each IPE cabinet (Class B rating) with not more than 3 cards per superloop
— Succession CSE 1000 Release 1.1	— No limit (Class A rating)
— Succession CSE 1000 Release 2.0	— No limit (Class A rating) of cards for each cabinet but it is recommended that no more than 4 Succession Media Cards in each Media Gateway since other slots are needed for PSTN trunking, etc.

Table 29
System IP Line capacity (Part 3 of 3)

Parameter	Capacity
Internet Telephone on each Voice Gateway Media Card	
— ITG-P Line Card	Maximum of 96 telephones supported on each ITG-P Line Card.
— Succession Media Card	Maximum of 128 telephones supported on each Succession Media Card.
Gateway ports on each Voice Gateway Media Card	
— ITG-P Line Card	— Maximum of 24 IP to circuit-switched gateway ports on each card.
— Succession Media Card	— Maximum of 32 IP to circuit-switched gateway ports on each card.
Internet Telephones in the System	
— Option 11C or 11C-Mini	— 640
— CP3	— 1000
— CP4	— 1000
— CPP	— 2000
— Succession CSE 1000 Rel 1.0 and Succession CSE 1000 Rel 1.1	— 640
— Succession CSE 1000 Rel 2.0	— 1000

Capacity engineering considerations

Number of Internet Telephones in the system

- **Option 11C or 11C-Mini** - There is a maximum of 640 Internet Telephones for an Option 11C or 11C-Mini. A maximum of five virtual superloops, 96-112 with cards 61-80 (640 telephones).
- **Option 51C, 61C, 81C, and 81C CPPII** - The number of Internet Telephones is determined by the engineering of real time usage, traffic capacity, network loop usage, and IPE slot usage, up to the maximum stated in Table 29 on page 121 for the specific CPU type.
- **Succession CSE 1000 Release 1.1** - There is a maximum of 640 Internet Telephones.
 - For normal traffic engineering, provision up to 1024 virtual TNs for each virtual superloop.
 - For a non-blocking virtual superloop configuration, do not exceed 120 virtual TNs for each virtual superloop.
 - A maximum of five virtual superloops, 96-112 with cards 61-80 (640 Internet Telephones).

Note 1: In Option 51C/61C/81C/81Cs, virtual superloops contend for the same range of loops with phantom, DECT, standard, and remote superloops, digital trunk loops and all service loops.

Note 2: Virtual superloops, phantom superloops, and DECT superloops contend for the same five superloops in Option 11C/11C-Mini.

- **Succession CSE 1000 Release 2.0** - There is a maximum of 1000 Internet Telephones. Up to 1024 VTNs can be configured on a single virtual superloop for Succession CSE 1000 Rel 2.0. Table 8 on page 47 describes the virtual superloop and virtual card mapping on a Succession CSE 1000 Rel 2.0 system.

For more information on configuring virtual superloops, see "Configure virtual superloops for Internet Telephones (LD 97)" on page 259.

Maximum number of Voice Gateway Media Cards in the system

Table 29 on page 121 describes the number of ITG-P Line Cards and Succession Media Cards that can be installed in a system.

Other Considerations

On a traditional telephone, the tones are generated by Meridian 1 and Succession CSE 1000. The Internet Telephone can generate tones that originate on the original switch, so the tones are not affected by distortion caused by compression codecs such as G.729A.

Traffic capacity of Voice Gateway Media Cards when supporting Internet Telephones

Each ITG-P card has 24 ports that are used for establishing a voice connection between Internet Telephones and non-Internet Telephones (such as digital telephones or public network). To configure a system as non-blocking (as is typically the case for ACD configurations), ensure only 24 Internet Telephones are registered on each card.

Each Succession Media card has 32 ports that can be used for establishing a voice connection between Internet Telephones and non-Internet Telephones. To configure a system as non-blocking (as is typically the case for ACD configurations), ensure that only 32 Internet Telephones are registered on each card.

A registered telephone is not synonymous with a configured telephone. When a telephone is registered, it is as if the telephone is plugged in. When the telephone de-registers, it is as though the telephone is unplugged. Registration consists of two steps:

- 1 Verifying the user's TN is valid and has not yet been registered.
- 2 Associating the TN on the Meridian 1 and Succession CSE 1000 side.

If an Internet Telephone is unplugged, it is automatically un-registered after a pre-determined time-out. This limitation on simultaneous calls depends on the number and type of calls (not on the number of ports).

A call between two Internet Telephones on the same Meridian 1 or Succession CSE 1000 IP Telephony node does not use the Voice Gateway Media Card as a voice path across the data network.

Voice Gateway Media Cards in a Meridian 1 and Succession CSE 1000 are pooled by customer number, are assigned dynamically, and are allocated preferentially by matching bandwidth management zones. For more details, see "VoIP bandwidth management zones" on page 164. An Internet Telephone can be assigned any port of any Voice Gateway Media Card within the Meridian 1 and Succession CSE 1000 system.

Note: The average number of Busy Hour Call Attempts must not exceed an average of 1200 BHCA each hour.

Refer to the following three examples for further clarification:

Example 1:

150 Internet Telephones with "typical" business usage of 600 call seconds per hour (CCS) for each telephone on average (for example, 5 calls of 120 seconds duration per hour)

- $150 \times 6 \text{ CCS} = 900 \text{ CCS}$
- 2 ITG-P Line Cards or 2 Succession Media Cards are required. Refer to Table 30 on page 129:
 - 2 ITG-P Line Cards support up to 1232 CCS
 - 2 Succession Media Card support up to 1738 CCS

Example 2:

500 telephones with "heavy" business usage of 12 CCS for each telephone on average (for example, 6-7 calls of 180 seconds duration every hour)

- $500 \times 12 \text{ CCS} = 6000 \text{ CCS}$
- 8 ITG-P Line Cards or 6 Succession Media Cards are required. Refer to Table 30 on page 129:
 - 8 ITG-P Line Cards support up to 6013 CCS
 - 6 Succession Media Cards support up to 6013 CCS

Example 3:

48 Call Center Agents with an allocation of 36 CCS for each telephone

- 2 ITG-P Line Cards or 2 Succession Media Cards are required
 - 48 ports required / 24 ports for each ITG-P Line Card
= 2 ITG-P Line Cards
 - 48 ports required / 32 ports for each Succession Media Card
= 2 Succession Media Cards (1.5 must be rounded up to 2)

Note: For Call Center Agents, it is recommended that one port be provisioned for each agent.

ISM parameters

Customers must purchase one Internet Telephone ISM parameter for each Internet Telephone installed on Meridian 1 and Succession CSE 1000 systems. A new ISM parameter uses the existing Meridian 1 and Succession CSE 1000 keycode to enable the Internet Telephone in the system software. The default is zero.

For a Meridian 1 system, the required ISM parameter depends on the system configuration:

- NTZC82AA Internet Telephone Software Parameter (Option 51C, 61C, 81C, and 81C CP II System)
- NTZC84AA Internet Telephone Software Parameter (Option 11C/11C-Mini System)

For a Succession CSE 1000 system, the required ISM parameter depends on the system configuration:

- NTM450AA Basic
- NTM451AA Advanced
- NTM452AA Premium

If you expand the ISM limits for the Internet Telephones, you must order and install a new Meridian 1 or Succession CSE 1000 keycode. Refer to the Incremental Software Management feature module in the *Features and Services* (553-3001-306) NTP.

Note: Individual ISMs are not supported on Functional Pricing. With Functional Pricing, ISMs are provisioned in blocks of eight.

Internet Telephone Engineering**Traffic and Service Circuits**

Virtual loops use software resources for tracking speech path traffic usage and Call Detail Recording. There are 120 of these resources for each virtual loop. The engineering of Internet Telephones is similar to that for existing digital telephones (based on 3500 CCS for each virtual loop).

The Voice Gateway Media Card's gateway channels are engineered the same as trunks between the circuit-switching fabric and the IP network. The TDS/Conference circuits are engineered for Internet Telephones as well as for existing digital telephones (one TDS/CONF card for each half group of Internet Telephones).

Gateway Channels Traffic Engineering

Configure no more than five Voice Gateway Media Cards on each superloop to eliminate the possibility of blocking because of insufficient talkslots (for example, 5 Voice Gateway Media Cards x 24 ports = 120 talkslots). Use Table 30 to determine the number of Voice Gateway Media Cards required to maintain the recommended capacity.

Table 30
Voice Gateway Media Card recommendations based on CCS capacity
(Part 1 of 2)

The Internet Telephone blocking probability is 0.005.		
Number of Cards	ITG-P Line Card CCS Capacity	Succession Media Card CCS Capacity
1	511	744
2	1232	1738
3	1996	2780
4	2780	3845
5	3577	4924
6	4383	6013
7	5196	7110
8	6013	8212
9	6835	9318
10	7660	10429
11	8488	11542
12	9318	12658
13	10144	13777
14	10983	14897
15	11818	16020

Table 30
Voice Gateway Media Card recommendations based on CCS capacity
(Part 2 of 2)

The Internet Telephone blocking probability is 0.005.		
Number of Cards	ITG-P Line Card CCS Capacity	Succession Media Card CCS Capacity
16	12657	17144
17	13496	18269
18	14335	19396
19	15177	20524
20	16020	21653

Note 1: CCS is the number of hundred call seconds per hour.

Note 2: If the number of ITG-P Line Cards exceeds 20, add 801 CCS to the total capacity for each additional card.

Note 3: If the number of Succession Media Cards exceeds 20, add 1082 CCS to the total capacity for each additional card.

Real time factors

The real time factors for Internet Telephones are provided in Table 31.

Table 31
Real time factors for Internet Telephones

Call scenario	Real time Factors for ITG-P Line Card	Real time Factors for Succession Media Card
1 - way inbound	0.78	1.75
1 - way outbound	1.59	1.75
2 - way	2.95	2.00

The total real time capacity of the Meridian 1 and Succession CSE 1000 depends on factors such as:

- calling patterns
- feature operations
- telephone and trunk signaling
- system CPU capacity

These factors are used to provision the maximum number of Internet Telephones supported on specific Meridian 1 and Succession CSE 1000 systems. These factors also describe the impact of using Internet Telephones relative to real time usage for a basic call between two 2500 telephones.

For Meridian 1 and Succession CSE 1000 Rel 1.0 and 1.1 systems, refer to *Capacity Engineering* (553-3001-149) for further information. For Succession CSE 1000 Rel 2.0 systems, refer to *Planning and Engineering Guidelines* (553-3023-102) for further information.

Product compatibility with other IP products

Nortel Networks supports the following Voice Over IP (VoIP) products in addition to the IP Line. These products are:

- Meridian 1 Internet Telephony Gateway Trunk 2.0 card/ISDN Signaling Link
- Meridian 1 Internet Telephony Gateway Trunk 2.1 card/ISDN Signaling Link
- 802.11 Wireless IP Gateway

Each IP product uses TLANs and ELANs that can co-exist with each other. All cards within a node must be on the same TLAN subnet. They can share the same TLANs, and must share the same ELAN. You need to engineer the traffic on the TLAN to consider all IP applications.

For EMC compliance, add up all the IP products to stay within EMC limits.